Lecture 14

Dancing Segway and Analysis of Musical Signal

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In this lecture, we will introduce Lab 6 and also some of the ideas behind the challenges, which will be published next week.

This lecture will provide some basic background for these lab sessions. In particular, you will learn how to detect beat and handle real-time data capture.

Segway Challenge – Aim and Objective

- To demonstrate your understanding of four topics in the Electronics 2 modules that are important to a design engineer:
  1. Signal processing;
  2. System analysis and design;
  3. Feedback control;
  4. Real-time embedded system

- The various challenges are designed to achieve the following:
  1. Apply what you have learned in this module to a real-life problem;
  2. Learn to combine offline processing using Matlab with real-time processing using MicroPython;
  3. Apply embedded system concepts and techniques such as sampling, buffer, interrupts, scheduling etc.;
  4. Have fun!

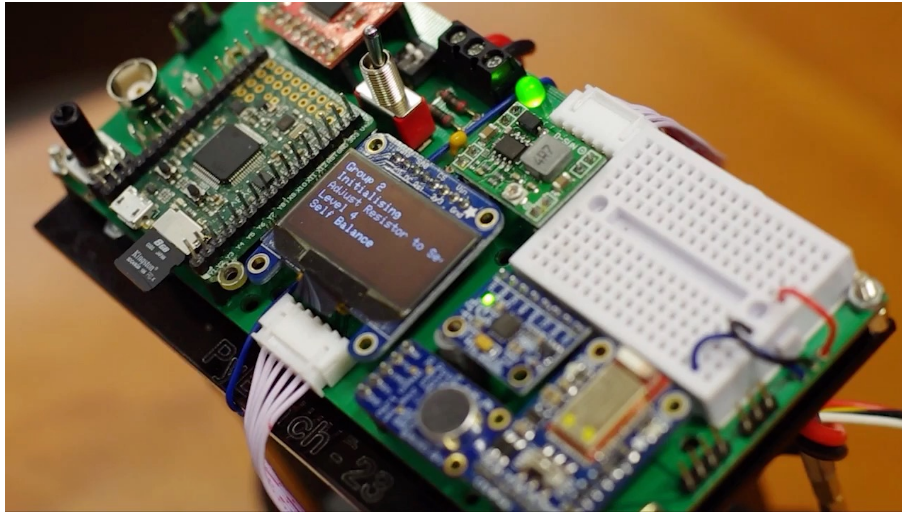
This slide shows the aims and objectives of the challenges outside the six laboratory sessions. So far, I have covered the necessary theory for aims 1, 2 and 4, either through lectures or through the lab experiments. I will be covering the topic of feedback control starting next lecture.

Segway Project – Learning Outcomes

- By the end of the challenges, you will be able to do most if not all of these:
  1. Process music signals using signal processing techniques to extract its signal characteristics such as rhythm (e.g. beat), ~~spectral contents (e.g. colour) and mood (e.g. swinging, loud, quiet);~~
  2. Creatively map the music characteristics to dance routines (manual);
  3. Analyse music signals in real-time on the microcontroller to synchronize dance movement to music;
  4. Balance a mini-Segway using a PID controller so that it moves around on two wheels under the control of your phone;
  5. Implement the mini-Segway that autonomously dance to live music.

For any module you take on your course, you should be very clear about learning outcomes. Here is the project's learning outcomes – stating what you should be able to do when you have finished this project.

Electronics 2 – from the past!



This is a video showing different groups achievement from a previous year. Some Segways only perform dance to music. Others can do dance and balance at the same time. The final clip shows driving a self-balancing Segway with a camera onboard!

Capturing real-time audio samples

- Sampling at 8kHz – assume that music signal under 4kHz
- Should use anti-aliasing filter (but not on PyBench)
- To capture the audio signal, you need to:
  1. Set up a timer to produce an interrupt every 125 microsecond
  2. Capture a microphone sample and put it into a buffer s_buf (i.e. an array) which stores N samples in sequence (N is 160 in my code, but can be changed)
  3. When the buffer is full (i.e. N samples capture), set buffer_full to TRUE (this is called a semaphore or a flag)

I now want to introduce you to Lab 6. The goal for Wednesday's lab is to capture audio data in real-time using interrupts, so that you can perform beat detection. Eventually you will apply what you learn from Lab 6 to synchronize your dance moves to the beats of the music.

You will also explore FIR filter (moving average type only), and to drive the Neopixel array to have dancing light with live music!

Setting up the Timer to generate an interrupt

- The microcontroller used on Pybench has many timers which can be programmed to produce interrupts
- We will use Timer 7 to generate the sampling interrupt
- Our interrupt service routine (ISR) is **isr_sampling**

```
# Create timer interrupt - one every 1/8000 sec or 125 usec
sample_timer = pyb.Timer(7, freq=8000)
sample_timer.callback(isr_sampling)
```


This is how you can program a timer (Timer 7) to produce an interrupt every 125 microseconds. The interrupt service routine is specified with the callback function.

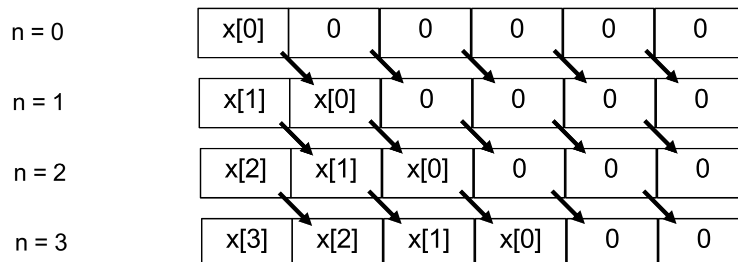
Ask yourself this question: what does the microprocessor do when an interrupt occurs?

The answer is:

1. Assuming the microprocessor is enabled to respond to interrupts, it will complete the current instruction;
2. It stores away (in some memory called "stack") the location of the next instruction, so that it can return later to continue your main program;
3. It stores away the internal values of the processor (called processor state or contex);
4. It jumps to the interrupt service routine and do whatever that specifies;
5. On completing the ISR, it recovers the state of the processor (or contex);
6. It return to the place in the main program where it was previously interrupted.

Buffering of signals

- In all the algorithms considered so far, we need to store N data samples. Data could be input music signal (from microphone) x[n], or instantaneous energy $\rho[n]$.
- In Matlab, this is easy. Matlab perform analysis offline, and you can store the signal is a huge array.
- In real-time system, this is not practical (nor possible!).
- Solution: implement a buffer:

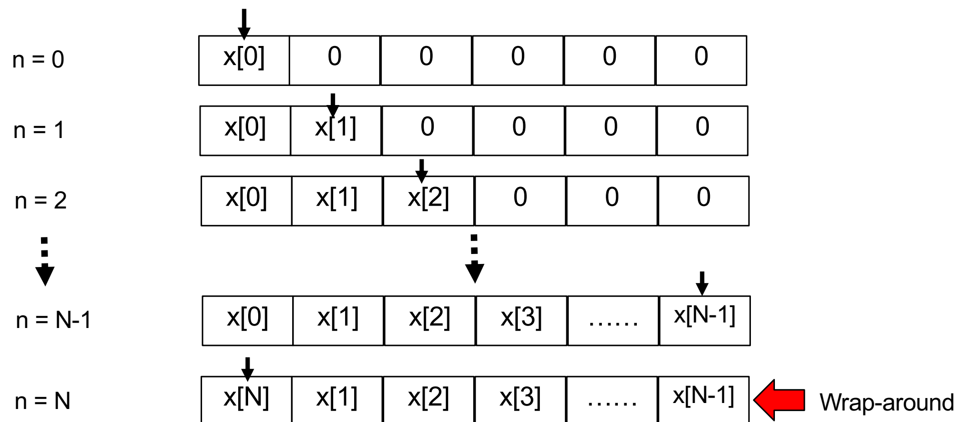


We need to take N samples of audio data in order to work out when a beat occurs. To do that, we use something called a buffer.

There is a type of buffer known as FIFO: first-in first-out buffer. This is shown in the diagram below. This is like a queue – queue to get onto a bus.

Efficient Buffering Method

- Instead of moving lots of data, you can use a “pointer” to specify where to put the new data:
- Use x[ptr], and increment ptr each time a new data comes in.
- Wraparound to 0 when ptr reaches N: `ptr = (ptr + 1) % N`



Using a FIFO is not efficient because you are moving data around a lot – that costs time and energy.

A better method is to use a concern call pointer. This is shown in the diagram above. You fill an area of memory (or array) in sequence, and move the the pointer (shown as arrow) up a position as shown above.

Note that if you keep incrementing the pointer, it wraps around at the end back to the beginning in a circular way. This is known as a circular buffer.

Interrupt Service Routine - isr_sampling

- The ISR do the following:
  1. Read microphone data
  2. Store it in the next location in array s_buf [ptr] – ptr is the index to the array
  3. Increment index by 1
  4. If index reaches N, buffer is full – set the flag (semaphore)

```
# Interrupt service routine to fill sample buffer s_buf
def isr_sampling(dummy):     # timer interrupt at 8kHz
    global ptr               # need to make ptr visible in here
    global buffer_full  # need to make buffer_filled visible in here

    s_buf[ptr] = mic.read()     # take a sample every timer interrupt
    ptr += 1
    if (ptr == N):
        ptr = 0
        buffer_full = True
```


Now let us consider how we write code in MicroPython to achieve all these. You will be doing it tomorrow in the Lab.

The interrupt service routine (ISR) is isr_sampling. We have to make ptr and buffer_full global variables because we need to access these OUTSIDE the routine. s_buf is also global – defined OUTSIDE the ISR.

The rest of the code is simple.

Beat detection using instantaneous energy (method 1)

- Assuming that sampling frequency is 8kHz
- We keep the current sample and N-1 previous samples of input x[n]
- Compute instantaneous energy of sound signal x[n] in, say, 20 msec window (N = 160):

$$\rho[n] = \sum_{k=0}^{159} x[n-k]^2$$

- One approach is to take the Fourier transform of the energy signal ρ[n].
- Collect 1-2 second worth (i.e. 50 to 100 ρ [n] values) and perform FFT on Matlab.
- The fundamental frequency of the spectrum ρ [jω] provides an estimate of the beat frequency.

Now we will consider three methods in determining when a beat occurs. The simplest way is the calculate the instantaneous energy of the sound signal. We have done this before. Once you have calculated the instantaneous energy for every 20 msec (160 samples at 8kHz sampling frequency), we can work out the periodicity of this energy to obtain the beat frequency (i.e. beats/minute).

In addition, you may also find the spectrum of x[n]. This gives you some information about the "colour" of the music. However, I found that it is rather difficult to deduce characteristic of music from the spectrum.

Beat detection using instantaneous energy (method 2)

- Compute instantaneous energy of sound signal x[n] in 20 msec window:

$$\rho[n] = \sum_{k=0}^{159} x[n-k]^2$$

- Compute steady state local energy by averaging 100 instantaneous energy values $\rho[0]$ to $\rho[99]$:

$$< \rho > \approx \frac{1}{100} \sum_{j=0}^{99} \rho[n-j]$$

- Beat occurs in the window when $\rho[n] > b \times < \rho >$, where b is a threshold chosen for the music.
- Method useful for real-time synchronisation (running MicroPython on Pybench).

Second method is an improvement. Apart from computing the instantaneous energy, you can also compute the steady state energy by averaging 100 instantaneous energy readings. A beat is deemed to have occurred if determining the energy rises above a threshold.

Beat detection using instantaneous energy (method 3)

- The problem of the previous method is that if you choose the wrong value for b, the algorithm will not work well.
- The threshold b need to adapt to the music itself. How?
- Compute the variance $v[n]$ of the instantaneous energy $\rho[n]$ over 20msec window:

$$v[n] = \frac{1}{100}\sum_{j=0}^{99}(\rho[n-j] - <\rho>)^2$$

- Now computer the threshold value b as:

$$b = \beta - \alpha \times v[n]$$

and try $\beta = 1.5$, and $\alpha = 0.0025$

Finally, you can even adapt the threshold value b according the music itself by computing the variance of the instantaneous energy. This is shown above.

Beat detection using Frequency selected energy

- Algorithm so far does not consider the frequency content of the music sound. That is, we ignore the frequency spectrum of the signal – it is colour blind!
- We know that beat information in a signal is actually frequency band related.
- Beat from drums – low frequency; beat from cymbal or triangle – high frequency.
- Therefore, assuming that our music is drum heavy, you can low pass filter the signal first before performing the previous beat detection algorithm.

Finally, you could include the frequency spectrum information in order to determine the beat. However, I don't recommend you doing this on MicroPython and Pybench board. This is because uPy is not fast, and our challenges involve three different things:

1. Capture music in real-time and extract beat;
2. Drive motors to dance to the music;
3. Control the Segway so that it self balance.

There is just not enough clock cycles left to do a FFT with the current hardware setup.

Package to drive motors

- The package **motor.py** is available to help you drive the two motors with ease. It will make developing your milestone code much easier.
- You must first import the package, and then create the motor object:

```
1  from motor import DRIVE
2  # create motor object for the two motors
3  motor = DRIVE()
```

- Thereafter, you can use the following methods:
- The first five methods are useful to control speed of the motors using the CONTROL PAD via Bluetooth
- The last six methods are directly controlling the movements of the two motors (in an open-loop manner)
- v is not really the speed, but the PWM drive value to the motors.

Method	Description
motor.up_Aspeed(v)	increase motor A speed by v
motor.up_Bspeed(v)	increase motor B speed by v
motor.dn_Aspeed(v)	Reduce motor A speed by v
motor.dn_Bspeed(v)	Reduce motor B speed by v
motor.drive( )	Drive motors at their set speeds
motor.A_forward(v)	Drive motor A forward at v
motor.B_forward(v)	Drive motor B forward at v
motor.A_back(v)	Drive motor A backward at v
motor.B_back(v)	Drive motor B backward at v
motor.A_stop( )	Stop motor A
motor.B_stop( )	Stop motor B

Since motor coils are essentially inductors, they have low DC impedances (resistance of the wiring). Hence when driving motors, we need to use special driver chips.

The driver chip you used in Lab 5 (the TB6612) is often called the H-Bridge Driver. Shown here is the simplified block diagram. There are four transistors connected to the supply rail and ground. (It doesn't matter which is which because the circuit is symmetrical.) The motor is connected in the middle forming the horizontal link of the H. The transistors are MOSFETs (metal oxide silicon field effect transistors) which is acting like a voltage controlled switch. When a '1' or high voltage is applied to the gate control terminal, the transistor turns ON and conduct electricity. If a '0' or low voltage is applied, the transistor is OFF. So the top diagram shows a configuration that results in the supply voltage being applied to the left terminal of the motor. The right terminal of the motor is grounded, and the motor turns in one direction. Reversing the control to the transistors results in the motor turning in the other direction.

If you use an AND gate at the control input, you can also add a PWM signal to control the speed of the motor.

Basically the '1' and '0' control signals are the A0 and A1 signals on the TB6612. The PWM signal is what you apply to the input of the AND gate.

Now you know how the TB6612 works.