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MiniMoog Additive Synthesis

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Real-Time System course**

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Hint

This system is developed under Linux Ubuntu 16.04 using Low Latency Kernel 4.8.0-36, Allegro5 for GUI and pthread library for real time periodic task managements

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1 System Architecture Overview

In this Document, we present an emulation of an additive synthesizer (the classical Mini Moog). In this system we relay on the **pthread** library for multi-threading programming. The system is responsible for the generation of the following wave-forms (sine, triangle and square) each with a specific frequency notes decided by users and then they are added/integrated together after they are passing through a dedicated Band Pass Filter (BPF).

Below in Figure 1, is the task models description. In the model multiple threads are running concurrently

1. TASK Graphics: allocated for rendering the graphics.
2. TASK Keyboard: Actively poll the keyboard to trigger and control the frequency notes of each wave
3. TASK Wave-forms: each waveform has an independent periodic tasks.
4. TASK Audio handles the Integration of the wave-forms and passing the generated buffer through the audio card driver.

The user can decide which frequency notes trigger for each waveform but Its limited to [Notes A4, B4, C4, D4, E4, F4, G4]. Also it's possible to trigger each wave to either ON or OFF and the same for filter.

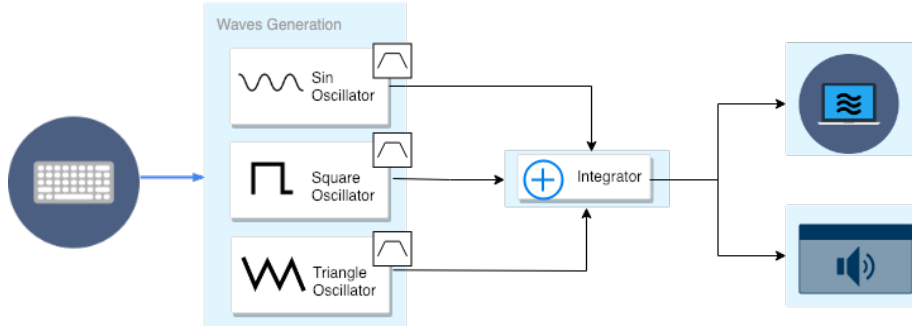


Figure 1: Additive Synths Task Model

2 Periodic Tasks Description

As illustrated below, 5 periodic tasks are running concurrently using the scheduling policy Round Robin [SCHED_RR](#). The main functions which are designed to manage periodic tasks are developed over pthread library **ptask** and **timemng** as reported in [\[3\]](#). Tasks are configurable through the file **task_configs**

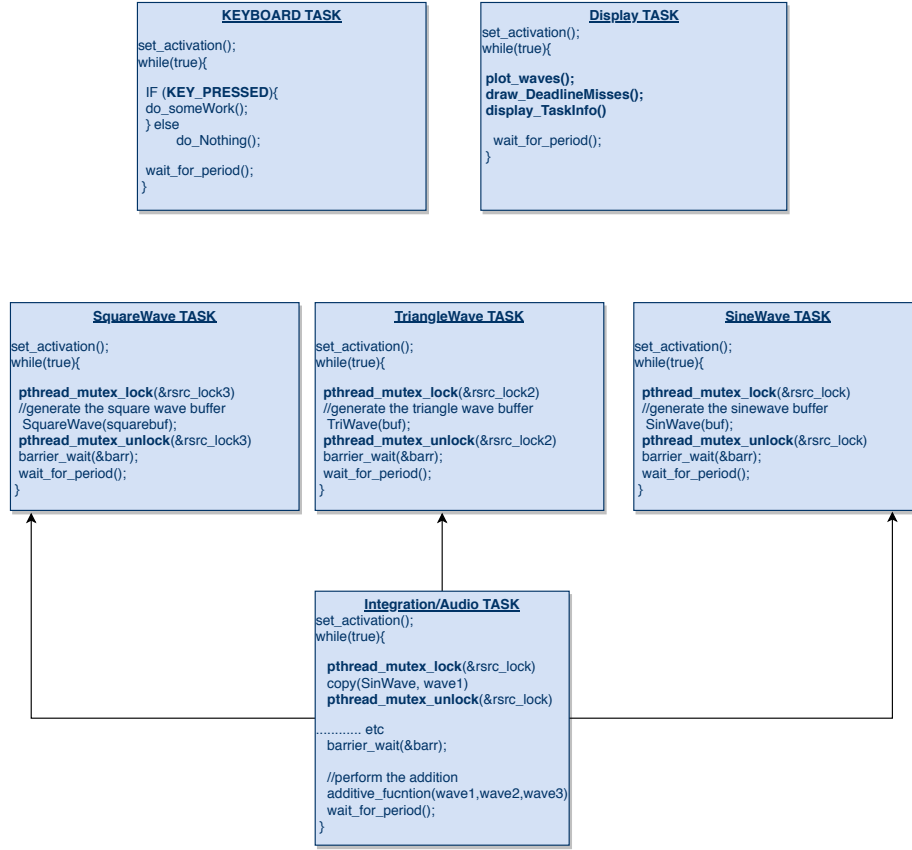


Figure 2: Periodic Tasks Model Description

The Inter-Task Communications between the Integration/Audio TASK which act as a reader and the Waveforms TASKS which are acting as writers are protected from simultaneous accesses to shared resources. This achieved through the mutex mechanism, which uses a semaphore variable of type **pthread_mutex_t** mutex semaphore to execute the critical sections of code in mutual exclusion. If our interest to have a very clear sound its necessary to provide a synchronization mechanism beside the mutex semaphore, otherwise, we can get a clear wave but sound will be a little bit distorted after each period, this is achieved by using barrier synchronization by specifying the number of threads that are synchro-

nizing on the barrier (4 in our case, 3 writers, one reader), and set up threads to perform tasks and wait at the barrier until all the threads reach the barrier. When the last thread arrives at the barrier, all the threads resume execution. However, a trade-off of using the barrier mechanism is the observed delay which in turns leads to deadline misses. Therefore, a **BARRIER_ENABLE** flag could be set 1 or 0 through task_configs file to allow or deny synchronization.

Reported below the tested configuration for all the threads in the system. And In the next sections each Task will be discussed in details

Table 1: Tasks Configurations

TASKS	ID	PERIOD	DEADLINE	PRIORITY
TASK sin_src	0	20	20	10
TASK square_src	1	20	20	10
TASK triangle_src	2	20	20	10
TASK audio_thread	3	20	20	20
TASK allegro_draw	4	50	50	30
TASK keyboard_thread	5	10	10	40

3 Digital Signal Processing Module

The waveform generation of the system was generated based on the standard *sin* function which is declared in *math.h* library. The produced samples of the *sin* function are 32-bit floating point so no the need to scale the produced output as the produced sampled goes in range between -1 and +1. Therefore all generated wave-forms streams that outputs sine, square, and triangle waves will be derived through to the sound card over the audio driver which could be configurable to be either *directsound*, *alsa*, *openal*, *oss* or *pulseaudio*.

All audio and waves declaration could be found in the *signal_src.c* and *signal_src.h* module there are 3 functions to generate the desired wave but all of them actually depends on the standard *sin* function.

1. **sin_osc**: generate a simple sine wave generated in the *sinWave_buf* which is self explanatory in the code.
2. **square_osc**: generating any wave could be done by adding different harmonic of *sin* wave. but in this case for simplicity of design, a simple three-hold was defined as a reference for the *sin* wave, if the three-hold more than 0 we generate a positive step (an amplitude of 1), and if its ≤ 0 we generate a -ve step. In this case there is no transition delay in falling or rising edges which is not practical but fits in our design which leads to an ideal square wave, samples were stored in *squareWave_buf*.
3. **triangle_osc**: triangle wave is generated using a similar approach from the *sin* wave using the *fmod* function, so we incrementally build the triangle wave using a linear step, samples were stored in *triangleWave_buf*.

Below as shown in **Figure 3**, is the schematic showing the process of the waveform generation, then integrating them into a single buffer and after that they derived using *alsa* driver to be played through the audio card.

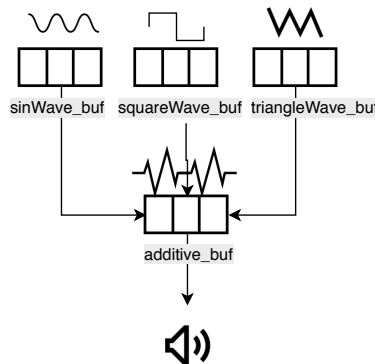


Figure 3: Audio driven flow and process

In order to make a precise sound output we have to set the output of sampling time of each wave to be equivalent to the audio sampling rate. In this case, where the audio sampling frequency could be configured to be 32 KHz, 44.1 KHz or 48 KHz therefore, the dt: **sampling time** should be”

$$dt = \frac{1}{FS_Audio}$$

Regarding the audio driver, we could set it using the allegro configuration file allegro5.cfg or it's always better to set it through the api in the application, which is defined in the minimoog.c init function

```
al_set_config_value(al_get_system_config(), "audio", "driver", "alsa");
```

The most important function is:

```
al_create_audio_stream(fragment_count, samples, freq, Audio_depth, chan_conf)
```

where we encounter a big delay in the system which produce and incremental increasing with respect to deadline misses but produces an overall very good sound quality following the default configuration.

in this table below is a two trade-off with respect to quality of sound vs deadline misses. Hence, the less the sampling frequency, the lower audio quality.

Table 2: Tasks Configurations

KNOBS		METERS	
Samples	Frequency	DLM	SoundQuality
1024	44.1KHz	1	percise
512	44.1KHz	0	poor
256	44.1KHz	0	poor
1024	32.0KHz	1	less percise
512	32.0KHz	0	less percise
256	32.0KHz	0	less percise

The choice of fragment_count, [\[1\]](#) samples and frequency (FS_Audio) directly influences the audio delay. The delay in seconds can be expressed as:

$$delay = \frac{fragment_count * samples}{freq}$$

```
sample_size = al_get_channel_count(chan_conf) * al_get_audio_depth_size(depth)
samples = bytes_per_fragment / sample_size
```

Finally, a Finite Impulse Response (FIR), Band-pass filter is designed to attenuate the signals if its desired [\[4\]](#), hence, Filter function is called one time only for each wave generator to generate the filter coefficient before the activation of each periodic task. **As shown in Figure [4](#)**,

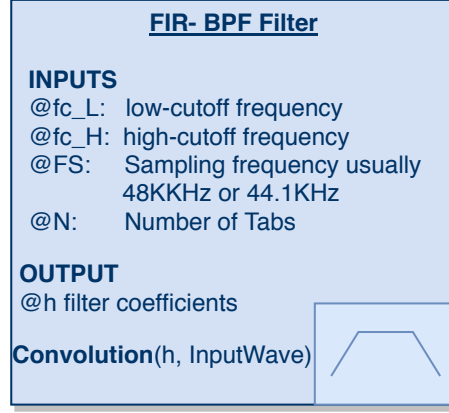


Figure 4: FIR, BPF

$$h = \frac{(\sin(fc_H * icount * PI) - \sin(fc_L * icount * PI)) / (icount * PI); _count * samples}{freq}$$

$$\forall icount = 1 : nCoeef$$

After computing the coefficient the signal is convoluted with the waveform buffer and then each waveform after passing through the BPF are added by a periodic task **Task Audio** which integrates the waveforms buffers after convolution and plays the audio stream. Hint the default setting is to play the pure sound waves without the filter, where filter activation is triggered by pressing SPACE key.

4 Keyboard Interpreter Module

The keyboard module is a periodic tasks does nothing except checking for pressed keys events. The user can control each waveform frequency note from A4 to G4 by pressing a corresponding key. Most of configuration are defined in the table and figure below. Hence, each keyboard row control a certain wave frequency in order A4, B4,... G4 in order from left to right.



Figure 5: keyboards Notes Map

Table 3: Keyboards Key Configurations

KEYS	FUNCTIONS
1	activate sinWave
2	activate squareWave
3	activate triangleWave
4	deactivate sinWave
5	deactivate triangleWave
6	deactivate triangle wave
-	Volume Up
+	Volume Down
A	sin Note A4, Freq 440.00 Hz
S	sin Note B4, Freq 493.88 Hz
D	sin Note C4, Freq 261.63 Hz
F	sin Note D4, Freq 293.66 Hz
G	sin Note E4, Freq 329.63 Hz
H	sin Note F4, Freq 349.23 Hz
J	sin Note G4, Freq 392.00 Hz
SPACE	FILTER ON/OFF

5 User Interface Module

TASK AUDIO ID: 3 Priority: 5 Deadline Miss: 0	TASK DISPLAY ID: 4 Priority: 6 Deadline Miss: 0	TASK KEYBOARD ID: 5 Priority: 20 Deadline Miss: 0	TASK WaveForms Sin DLM: 0 Square DLM: 0 Triangle DLM: 0
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Figure 6: Concurrent Task Information Field

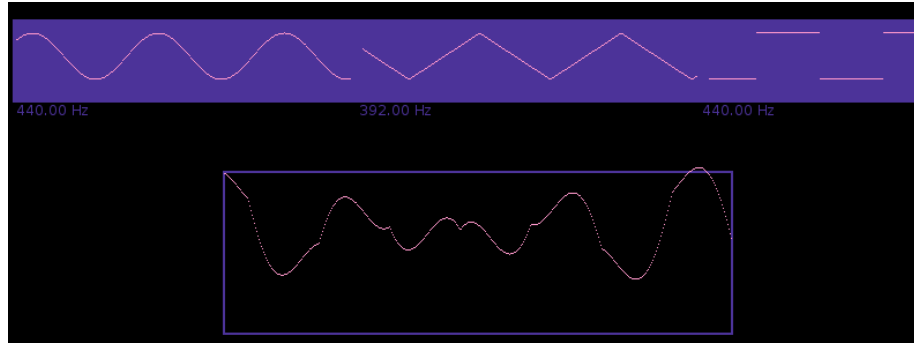


Figure 7: Oscilloscope Monitor

	SIN	TRIANGLE	SQUARE		SIN	SQUARE	TRIANGLE
NOTE A 440.00	A	Q	Z	ON	[1]	[2]	[3]
NOTE B 493.88	S	W	X	OFF	[4]	[5]	[6]
NOTE C 261.63	D	E	C				
NOTE D 293.66	F	R	V				
NOTE E 329.63	G	T	B				
NOTE F 349.23	H	Y	N				
NOTE G 392.00	J	U	M				

-----To toggle the filter press SPACE-----

-----Volume up (+)-----

-----Volume down (-)-----

Figure 8: Instruction Field

The Display is divided into 3 fields, Task related information, to monitor deadline misses. Oscilloscope monitor field which is responsible for plotting/rendering signals samples in real-time, hence each sample is represented by a pixel [2].

$$\forall i = 1 : Samples_Per_Buffer$$

$$al_draw_pixel(x, y, i)$$

where x and y are the screen coordinates

References

- [1] Allegro5 manual https://www.allegro.cc/manual/5/al_create_audio_stream. *Allegro 5.0 reference manual - Allegro 5 Manual*.
- [2] Allegro5 manual https://www.allegro.cc/manual/5/al_draw_pixel. *Allegro 5.0 reference manual - Allegro 5 Manual*.
- [3] G. Buttazzo and G. Lipari. Ptask: An educational c library for programming real-time systems on linux. *2013 IEEE 18th Conference on Emerging Technologies & Factory Automation (ETFA)*, 2013.
- [4] D. Morgan. *Practical DSP modeling, techniques, and programming in C*. Wiley, 1995.