**Organisational Note**

* You are not to touch this project unless you intend to add a geometric algebra based visualisation by the next version. You have spent enough time on administration.

**Introduction**

AudioVisual is an OpenGL/Audio project that has the intention of providing an interface between graphical models and audio streams. The objectives of this implementation are:

* **Audio** *(Meta: These are goals not features)*
  + Smooth, error-free, realtime, threaded access to an audio buffer.
  + Access to audio buffer by file, or by realtime input.
  + Accessible interface providing audio analysis metrics; including pitch tracking, energy analysis..
* **User Interface**
  + Modular implementation of the immediate-mode GUI
* **Visualisation Engine**
  + Simple graphics engine with: *lighting, AA, normal mapping, texture & model loading.*
  + Model-view based rendering
  + Action-queue framework to build behaviour.
  + A High-dimensional model, with a dimension-free polytope winding and interpolation algorithm.
* **Geometric Algebra**
  + An incorporation of an implementation of geometric algebra (say *versor*), into the structure of your visualisations.
* **Visualisations**
  + A set of visualisations, built within the bounds of the engine, with their own setting systems
    - Tetrahedra in a field
    - Transformations on a Tesseract
    - Normally distributed cubes
    - Fractals
* **Render Configuration**
  + The ability to run the application in two modes:
    - Realtime Mode: *View and configure audiovis from realtime buffers.*
    - Scripted Render Mode: *Configure audiovis in advance, and save output to a file*
* **Development Process**
  + Clean versioning: requirements, installer, changelogs, testing process, release process.
  + Information: description and how-to, installation instructions,

**Meta Notes**

* You should focus heavily on requirements and define them well.

**Versions**

* **On every version:**
  + Changes are logged in a changelog (mostly user-oriented)
  + An installer is made.
* **Choosing features for a version:**
  + Every version has at least one visible use-case modification, even if it is simple
  + Tick-Tock Approach:
    - Even version revisions (0.2,0.4..): are more focussed on end-user changes
    - Odd version revisions (0.1,0.3..) are more focussed on infrastructure changes

**Process**

**Scheduled Features (***Next Version:***0.3)**

* 1.a.ii.1 – Multibuffer
* 5.a.i - Render-to-file Part I
* 6.a. Unit Testing

**Urgent Features**

* 1.b.i – Audio Stream Settings

**Candidate Features**

1. **Audio:**
   1. **Buffer:**
      1. Audio Analysis Metrics: *Either: Import GIST correctly or Implement DFT, energy metrics etc yourself.*
      2. Long & Interpolated buffer:
         1. Multibuffer: *The user should be able to choose not only the sample size of the recorded buffer, but a suitable multiple (1-224), or duration of these samples to be stored in a longer term buffer. (Note: you should look up deques/queues for this). The multibuffer should be appendable. Bonus points: Later, when synchronisation is implemented, the multibuffer size, buffer size, interpolation rate, should be automatically optimised.*
         2. Buffer Interpolation: *When using audio for visualisation logic, a buffer should be obtainable from the multibuffer, with a certain offset from the end in ms or samples.*
      3. Timestamp Management: (Current problem is that
   2. **Stream**
      1. **Audio Stream Settings**: *Build an interface to the Recorder that allows you to (a) see all stream properties via a public interface, (b) Modify specific stream properties, re-initialising PortAudio safely with different settings: Channel No, Left/Right, Sample Size, HostAPI, Device*.
      2. **Audio Files:**
         1. Part I: Implementation: *Build an interface that can load an audio file and play it into a buffer of the same type. (.wav only for simplicity)*
         2. Part II: Abstraction: *The audio file stream should be an implementation of an abstract stream, sharing type information with the PortAudio recorder class.*
         3. Part III: User Interface: *Allow the Audio Mode to be swapped, and let the user choose an audio file from the interface.*
2. **User Interface**
   1. **IMGUI Renderer**: *A class which holds all window instances, and holds a render function with all the main IMGUI setup/shutdown calls.*
   2. **Window/Settings Design**: *A tree/list of windows each with their corresponding settings class. Windows are held as shared\_ptrs by the UserInterface class, and Settings are held as shared\_ptrs by the Program class. Interfaces ask the Program to retrieve their settings when they are drawn, or when they want to modify the settings. Unanswered: If I start recording or restart the audio interface for instance, what makes the call to do this? i.e. You might not want to ask the UI to do this directly, since then you rely on the UI for behaviour. You don’t want to ask the settings to do this because it is not its job. Hence MVC is probably the most suitable pattern for this problem statement.  
      Controllers should contain Settings objects and Windows should talk to Controllers to get them. Reasons are:  
       (a) UI is not responsible for behaviour and nor are settings or some global manager  
       (b) Interdependence between UI elements is handled by controllers. For instance if a setting   
       in the Video Rendering options is dependent on state in the Audio Input options, then (you need to  
       actually find a use case and decide what is responsible for this)  
       (c) The controllers can talk to each other by asking the Program object, and that way we can  
       use the Program object to handle thread-safety and synchronisation.  
       (d) The settings can be made accessible from the controller as const, and can be reset only in  
       their entirety. That way the complexity of use cases can be handled carefully by the behaviour  
       of the controller.  
      The Program instance, or a descendent of it, might do well to be the primary, private owner of settings object. Reasons are:  
       (a) the project is not dependent on UI to function, since Settings can initialised and modified without all of the controllers. This will be useful with Configuration based Rendering.  
      (Note: Actually look at MVC patterns and see if this is in the same order)*
      1. Part I : Create MVC setup for Video Rendering. The window asks the controller for the settings when drawing. When settings are changed the UI talks directly to the controller, so that the controller can make its own calls. Controller\_VideoRendering
      2. Part II :
   3. **Main Window**: *A window from which you can show/hide all other windows, itself toggleable with a hotkey.*
   4. **Audio Window**:
3. **Visualisation Engine**
   1. **View-Model System:**
   2. **Audio Attribute Mapper:** *(Choose visual attributes to be mapped to audio attributes with an interface. Interface should have the ability to cycle through permutations of mappings in a gradual fashion.)*
   3. **High-Dimensional Model**:
   4. **Visualisation Transition System**: *(Needs more definition but: Use commonality in vertex/shader info to smoothly move from one visualisation to another at a suitable time)*
   5. **Camera**
      1. Part I: Initial Implementation
      2. Part II:
   6. **Visualisation Base**
      1. Abstract Configuration Settings:
   7. **Scene:** (*The scene should hold and render all visualisations and manage OpenGL state. It should also mediate communication on a global level between non-UI OpenGL models and views*.
4. **Visualisations**
   1. **Tetrahedra in a Field:** *(Tetrahedra are generated with some thrust in a direction, from a rotating point in the middle. They orbit the centre, and are pulled in and pushed out from the centre. They shrink over time, and at a certain size threshold they disappear.)*
      1. **Map Spatial Properties:** (Space: *Generation Rate, Field Strength, Field Deformation*, Shapes: Velocity, Colour, Shrink Rate)
      2. **Governing Audio Properties:**
   2. **Exploding Cubes**
   3. **Julia Fractal:** (A 3D Julia fractal, whose properties change with parameters of the audio.)
   4. **Parametric Geometry**
   5. **Point Clouds**
   6. **Spectral Surface Skybox** (The time-domain or frequency-domain data mapped to a surrounding surface)
      1. Part I : Build a mesh over the time/freq-domain data.
      2. Part II : Build configurations for wrapping the mesh: Find a way to wrap it around an enclosing cylinder.
      3. Part III :
5. **Render Configuration**
   1. **Render-to-file:** *The ability to write the framebuffer to a file*.
      1. Stage I
         1. Part I : Initial Implementation: *Load example from (*[*https://stackoverflow.com/questions/3191978/how-to-use-glut-opengl-to-render-to-a-file*](https://stackoverflow.com/questions/3191978/how-to-use-glut-opengl-to-render-to-a-file)*) and build a vcxproject for it. Get libraries and produce includes & policy for it. Modularise it so that it is easy to incorporate to the main project.*
         2. Part II : Incorporate: *Incorporate the render-to-file code into the main project. Build an interface so that you can start & stop a render.*
         3. Part III : Flexibility: *Allow the user to choose the quality, the codec*
      2. Stage II
         1. Part I : Implement interleaved encoding of audio into the file.
         2. Part II : Implement settings to choose offset from
   2. Command-Line interface:
   3. Configuration XML files:
6. **Development Process**
   1. Implement Unit Testing for the main modules.
   2. Create installation instructions for github, with a list of dependencies.
   3. Make sure that all build configurations (x64 & x86 Debug/Release) work correctly.
   4. Create a publishable installer, with a policy to update the installer.

**Completed Features**

**Current Problems**

1. **Audio**
2. **Synchronisation**
   1. You need a way of synchronising audio and visual processing. For visualisation logic, an iteration through the logic must include: the time that has passed, the audio buffer to use for the iteration, the time travelled through the audio buffer
      1. **When Processing Visualisation Logic:** All calculations are referenced by the amount of samples that have passed.
      2. **When Getting Audio Buffer:**
      3. **When Rendering to Screen:**

**Unorganised Features**

1. You’ll want to add skyboxes. You might want to *animate* the skyboxes, and that would require them to exist as their own visualisation.
2. If you are doing audio processing in advance, and also moving mapped attributes along a ring of parameters for a visualisation, then:
   1. You’ll need to know the range of all of these properties in advance, to be able to normalise their contributions to each parameter. *Normalisation should be an option in* 
      1. For instance: if the pitch takes a value from 0.3-0.5 but the amplitude takes a value from 0.01-0.93, if they are both contributing to a property (say field strength), they need to be normalised so that the move in contribution is gradual.
3. Configuration system:
   1. You should have two options by default: Scripted & Realtime. Scripted is enabled if the program is ran with a “.xml” file.
   2. You should be able to generate a configuration from within the program.
4. UI & Settings:
   1. UI retrieves and modifies properties in settings classes
   2. Settings classes are read to determine behaviour in scene
5. Adding Visualisations together

**Unsorted Administrative TODO**

* Dependencies:
  + You must add FFMPEG bins for x86, and confirm that every configuration works correctly
  + At some point you might want to change the libraries include sheet so that you can group folders properly, by including each library as its own folder.