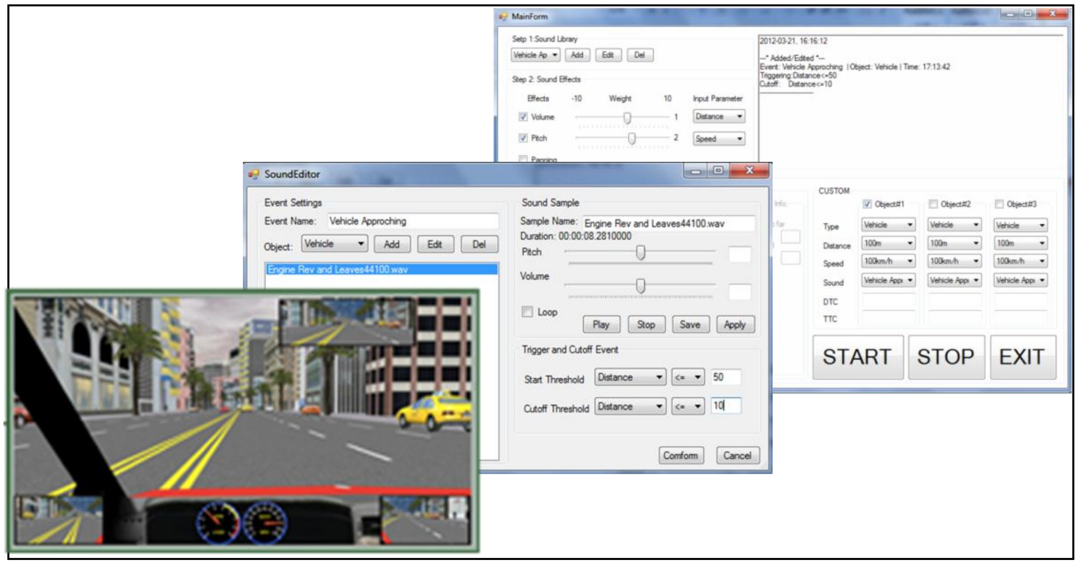

Development of an intuitive sound manipulation tool for in-vehicle human-machine interaction design

Master's Thesis in Intelligent System Design

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Abstract

Intuitive sounds such as auditory icons are known to be more efficient than abstract sounds (earcons) in conveying driver relevant information. Further, different traffic situations may be of different urgency levels and also related to the driver's performance. Hence auditory information may have to be changed or manipulated in order to convey the appropriate level of urgency. However, very few authors address the problem of appropriately designing auditory icons and how they should be manipulated to convey different urgency levels. This thesis work has been conducted in order to develop a signal processing tool which could be used for such design and evaluation. The tool is designed to take different sensory data (as distance to a leading vehicle) as the input and use that data to manipulate a catalogue of sound signals. The goal of the thesis is to let these sound signals inform the driver about the current traffic situation with the right level of urgency.

Keywords: Auditory Icons, Human Machine Interaction, Real-time Audio Manipulation, C# Programming

PREFACE

This thesis work has been carried out as part of the Environmental Friendly efficient Enjoyable and Safety Optimized Systems (EFESOS) project. The main project partners are Semcon AB and Chalmers University of Technology.

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I would like to thank my supervisor, Per-Göran Åkesson from Semcon AB, for offering this thesis work and his advices concerning software requirements and structure. I would also like to thank my examiner Professor Claes Strannegård from Chalmers University of Technology for his great inputs to the thesis work and report.

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Last but not the least; I would like thank my parents and all my friends. They have always been supportive. I am very grateful.

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Chapter 1 Introduction

1.1 Background

The technical evolution in today's computer and automotive industries has made it possible and attractive to have a variety of information systems in cars, thus making the driving experience safer and more enjoyable. For instance, there are GPS systems guiding the driver to a destination; advanced driver assistant systems (ADASs); and multimedia systems helping the driver fetch the latest news and so on. Nowadays, it is common that the driver interacts with numerous in-vehicle devices while driving. Some of these interactions would cause distractions and thus reduce driver's performance. Answering a phone, texting or reaching over to manage infotainment systems while driving, are some examples of driver distraction. Studies have shown that distractions from interactions requiring too much of the driver's visual attention could lead to serious accidents.

According to Wickens et al's (2002) MRT (Multiple Resource Theory), there is better time sharing performance if we present information through different information channels or modalities [1]. Human cognition incorporates a variety of sensors distributed over the human body which receives various inputs from the environment, but only three of them are commonly used in driving situation [8] - visual sensation followed by auditory sensation and tactile sensation. For now, almost all the interactions require the driver's visual attention. In order to avoid distractions, one solution is to use other two channels (auditory and tactile) to transfer information to the driver.

In fact, this concept has been used for vehicle infotainment system design. Auditory information systems are widely used nowadays. For instance, when reversing a car at a parking lot, sensors on the back detect the distance to obstacles behind the car. The infotainment system starts producing 'beeps' also called earcons and changes their pitch based on the distance to detected obstacles, helping the driver to react accordingly. However, studies have shown that the driving performance depends on the driver's learning and cognition abilities. When a new sound is produced, there is a risk that the driver would feel insecure and be distracted. After a learning process, the driver might get used to the sound, but every time he or she needs to think a little bit before reacting which would be undesirable.

An alternative method is to use speech-based interaction in cars. The system reads text messages, emails, and navigations using actual verbal communication. Some studies discovered that speech-based interaction works great with a less urgent event, but does not address active safety issues [11, 12]. When an urgent event occurs, the driver has to wait for the system to finish its sentence to know what is really happening, which might be dangerous.

Speech-based information system attempts to increase the information density to cope with the disadvantages of the abstract sounds, which could be distracting. Other ways of increasing the sound information density would be to use:

> –Directional cues, through 3D placement. For example, the research has shown that visual attention can be directed to critical events by using spatially informative sounds [2].
>
> –Time varying sounds. For example, the perceived urgency could be increased by increasing the pitch of a sound [3].
>
> –Sounds associated with the meaning of an object, i.e. auditory icons, are known to be more effective than abstract sounds in conveying traffic information. It could also reduce the driver's response time [4].

The auditory icons were defined as 'everyday sounds mapped to computer events by analogy with everyday sound-producing events' [5]. Researches has shown that the auditory icons were easy to learn and more comprehensible to drivers [14, 15, 16, 20]. Compared to speech messages, the auditory icons shortened the driver's response time in urgent situation [18, 19]. In additional, Chen (2007) indicated that the sounds associated with meanings were able to enhance the safety of the driver by warning them about surrounding traffic [17]. Since auditory icons are considered as a better way to inform drivers about traffic situations, there must be some studies to verify if auditory icons related to a road user are better than conventional sounds.

1.2 Purpose and structure of the thesis

The purpose of this thesis work is to design and produce sound manipulation software, which helps sound designers and HMI researchers to evaluate the performance of auditory icons. Sound designers from Semcon have produced a series of early warning sounds and want to test if they are acceptable to drivers before implementing them in vehicles. Therefore, certain software is needed that automatically communicates with a driving simulator, while tracking other road users, and triggers and manipulates the auditory icons.

The classic software development model, the waterfall process model [13], was followed in the thesis work, including problem definition, requirement analysis, software design, implementation and verification. The report follows this exact order to present the work.

Chapter 1 introduces the overall background and the problems that this software intends to overcome.

Chapter 2 analyzes the software requirements using a user-centered method. Results from interviews with target users are presented in this chapter and categorized into more structured requirement specifications.

Chapter 3 divides the software into four different functional models and explains the reason for such designs and how they are related.

Chapter 4 describes the problems and solutions when implementing the software. The last chapter presents some detailed software performance, including the software's efficiency, timeliness, and capacity to process multiple sounds.

At the end of the report, possible future work is discussed.

Chapter 2 Requirements Analysis

2.1 User requirements analysis

Since the software was user-centered, this meant that user requirements analysis was considered as a key element to define the software's structure and functionality. Users, in this context, were sound engineers from Semcon AB and HMI researchers from Chalmers. The purpose of this step was to extract the specific requirements from users' expectations. Laplante (2009) provided the following guidelines, a list of questions to answer, to help define the basic software requirements [10]:

1) *Operational distribution or deployment*: Where will the system be used?
2) *Mission profile or scenario*: How will the system accomplish its mission objective?
3) *Performance and related parameters*: What are the critical system parameters to accomplish the mission?
4) *Utilization environments*: How can the various system components be used?
5) *Effectiveness requirements*: How effective or efficient must the system be in performing its mission?
6) *Operational life cycle*: How long will the system be in use by the user?
7) *Environment*: What environments will the system be expected to operate in an effective manner?

The questions above were answered by conducting interviews and discussions with target users in the earlier stage of this thesis work. The main findings can be organized as follow:

1) Operational distribution or deployment:

   The software would be used for the real-time driving simulator software – STISIM, and for processing sounds offline.

2) Mission profile or scenario:

   The software was supposed to work between the traffic environment and the driver, which could be seen as a layer which extracts the traffic information and transfers it into more perceivable information format. In this context, this new information format is the auditory icons.

Therefore, a layer was required to receive the traffic data (the vehicle's speed, acceleration, position, etc.), analyze and understand the current traffic situation, and then trigger the related auditory icons. Moreover, this early warning system differed from the conventional systems, since the sound effects like pitch shift, spatial panning, etc. could be applied in real-time based on certain triggers.

3) Performance and related parameter

For all vehicle warning systems, the timeliness is known to be the most critical parameter to determine their performance. The shorter the response time it requires to perceive warnings, the better warning system it is. Any latency might lead to some serious dangers or injuries. Therefore, in the software programming point of view, the computing speed of the programming language and the efficiency of the code were considered as the most critical software parameters.

On the other hand, could not only the software's performance be evaluated by its non-latency triggering, but also the capacity of multiple sound effects and tracked vehicles. All these factors were related to the computing speed and well-structured code.

4) Utilization environments

At present, the utilization environment is the vehicle simulation room at Chalmers, instead of implementing in a real vehicle. The equipment and their connectivity are shown in Figure 2.1.

1. The driving equipment: a driver's seat and a control system including steering wheel, pedals and shifter;
2. The visualization system that displays the traffic simulation, both LED screen or projector are accepted;
3. Loudspeakers for playback of auditory icons;
4. Computer 1 running STISIM simulator software, which controls the driving equipment and the screen and passes the traffic information data to Computer 2;
5. Computer 2 running the test software, which receives data from computer 1 and plays the processed sounds using loudspeakers

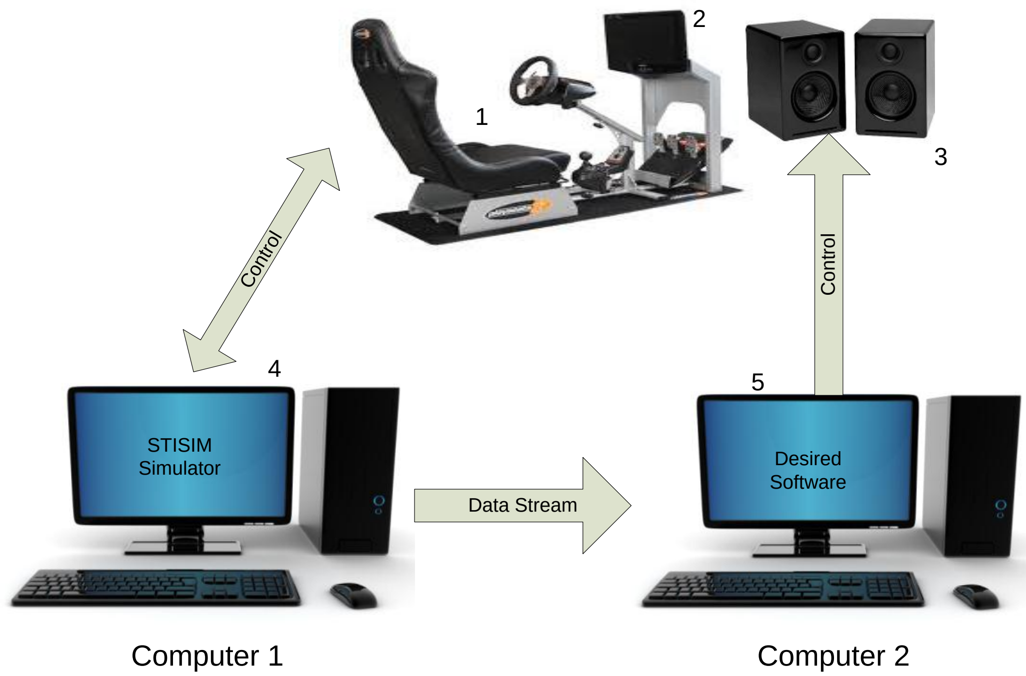


Figure 2.1: Software Utilization Environments

5) Operational life circle

The thesis work is an effort to improve the testing method for in-vehicle information systems. For now, the software only operates in a laboratory with the vehicle simulator, instead of a real vehicle. However, in the future, if the concept of using auditory icons for in-vehicle early warning system is verified as useful and feasible, further research could be done by developing this software. Thus it is important for the software to be open-sourced and well-structured.

6) Environments

Environments, in this context, address the operating system (OS) and the programming language (PL). There are many choices available in the market, but a desired environment should contain following properties:

- OS and PL are well known to target users, requiring less effort to learn.
- Easy to code, with the possibility to modify the code.
- PL is commonly used so that the user can access the HELP from anywhere like internet, technical support and etc.
- OS and PL are easy to maintain and develop in the future.

2.2 Programming language

In order to fulfill the needs of the software, the programming language should have outstanding properties concerning the computational speed, ease-of-coding, and flexibility for future development. The most popular programming languages nowadays, like MATLAB, C++ and C#, have been taken into consideration and passed through two feasibility tests to find out the most suitable one.

The first test intended to examine the computational speed for real-time sound activation and manipulation, which requires the programming language to be able to trigger a sound without delay and manipulate pitch without distortion.

MATLAB is widely used as a signal processing tool and has a collection of built-in signal processing functions, suitable for the sound manipulation. In the test, it is easy to trigger events using MATLAB functions. However, pitch shifting is virtually impossible to implement, because the built-in wave play function does not allow changing pitch during real-time playback.

To tackle this problem, the audio stream was cut into small clips. A user-defined function applied a pitch shift filter to each sound clip in advance, while the previous clip was played back. However, this method failed as well, because it was not fast enough to apply the filter before the last playback, resulting in gaps between successive sound clips.

C++ is considered as the most powerful programming language, especially for real time processing, and a third party application, 'Csound', was found to be useful. Also for C#, a third library, called 'NAudio', was an open source .NET audio library, containing dozens of useful audio related classes intended to improve audio related utilities in .NET. It has been in development since 2001 and has grown to include a wide variety of features. It has basic functions like audio playback, and also powerful audio mixing. Both C++ and C# had extraordinary performance on the first feasibility test, which means these two languages were compared further.

The second test was designed to find out the language's capacity of handling multiple sounds. Both C++ and C# worked perfectly with audio mixing. However, the biggest problem with C++ was that the Csound library was unable to communicate with STISIM. It needed an API (Application Programming Interface) to transfer the data stream, and this API required a Linux-based programming environment on Windows OS, which was hard to configure since Windows 7 had blocked such access.

The detailed comparison among these three programming languages is shown in Table 2.1. It is necessary to point out that GUI (Graphic User Interface) design is another important factor for choosing C# over others. A variety of GUI components from C# has extraordinary performance when it comes to multi-tasking, and is easy to learn and code as well.

Table 2.1 Programming language comparison

	C#	MATLAB	C++/Csound
Built-in functions or libraries concerning audio processing	Yes	Yes	Yes
Computational speed for real-time processing	Yes	No	Yes
Multi-thread programming	Yes	No	Yes
Object oriented programming	Yes	No	Yes
Easy to configure and code	Yes	Yes	Not really
Environment requirement	Win	Any	Linux like
GUI design	Great	Not Good	Good
Result	O	X	X

2.3 Software requirements specifications

2.3.1 Sound editing of auditory icons

The software should contain some basic functions which could be found in other audio editing applications, such as:

- Import sound file (WAV or MP3) from hard disk to software
- Manipulate sound parameters (volume and pitch)
- Save and delete modified sounds

Meanwhile, compared to other audio editing applications the sounds in this software represent traffic objects or events and the sound parameters are related to the urgency level. In addition, as a testing software, the sound designer needs to try out the different sounds, sound effects and triggering conditions to find the best combinations. In order to do this, the sound editing should be able to:

- Associate auditory icons with traffic objects or events
- User-define triggering conditions
- User-define the mechanisms of sound effects

2.3.2 Intuitive sound effects

As is known, the sounds with time-varying frequency contents deliver more information than the static frequency contents. Three sound effects are implemented in the present work - volume change of the sound, which would relate to the distance to objects; pitch shifting, which indicates the speed of objects; and spatial panning, which helps the driver to locate the traffic objects. The amplitude of the sound, for instance, is corresponding to the volume variation and its frequency is related to pitch shifting.

The sound effects are not going to be fixed but depend on the user’s needs. In this context, effect mechanisms become complex since the sound parameters are associated with traffic parameters (speed, distance, direction, etc.) and the range of each sound parameter is also unknown.

Therefore, it is important to leave flexibilities to users to configure their own sound effects, by setting up different effect mechanisms. There are two reasons for that. First, the traffic situation differs over time, it is impossible to use one fixed effect mechanism for every single event. Secondly, the testing software needs to try out different parameter settings and effects combinations. For instance, the effect, volume change, could be related to the distance change between two cars, but it also could be related to the relative speed. The Doppler Effect usually is defined by the relative speed and distance, but the weights of these two factors are unknown. Users might want to change the different weights of the speed and distance for a better solution. Therefore, it is better for users themselves to define the effect working mechanism:

- User defines what sound effect or effects to use in the test
- User defines connection between sound parameters and traffic parameters
- User defines the weight of each sound parameter

2.3.3 Traffic tracking and intuitive sound triggering

A warning system is not only activated when a critical event happens, but also able to predict the potential dangers. Sensory data is passed to the processor which determines urgency levels and corresponding actions. In this thesis, the software is considered as this processor. It receives the data from the driving simulator, determines the current situation, and triggers the appropriate auditory icons. Therefore, the software should be able to:

- Receive traffic environment inputs.
- Keep tracking traffic objects.
- Examine the risk of collision.
- Trigger the playback of auditory icons.

2.3.4 Graphical user interface and user controls

In the previous sections, the overall functionalities of the software were discussed. Its user-defined features provide flexibility for the end user but cause some challenges when designing the user interface. Nowadays, the most commonly used user interface (UI) is the Graphical User Interface (GUI). However, a good GUI is not that easy to design. If one considers the sound settings for the auditory icons, the operations could not be accomplished by one or two clicks. The lowest requirement for GUI would be an accepted usability that can guide users to accomplish their tasks even without a user manual. In this context, the GUI should be able to:

- Guide user how to set up test scenarios, imports sounds, defines sound effects, connects with simulators, etc.
- Make functional GUI components comprehensible by their appearance. For instance, a button named “Play” indicates audio playback.
- Give feedback whenever a user triggers an event, including the software status, notifications, and error messages and so on.
- Increase the efficiency of the software by reducing the number of inputs from the user as much as possible.

Another aspect of the GUI design is the user’s ability to control the software during testing. One solution to do this is restarting the test from the top, which is clearly not a good idea. An alternative solution would be if users could disable it as soon as they can. Therefore, a GUI is a good solution for real-time user controls, including:

- Enabling and disabling sounds during the test
- Applying and removing sound effects during the test
- Modifying the effect parameters and weights during the test

2.3.5 Open structure for future development

Open structure is a universal requirement for software development. It means that the software structure and codes should not constrain to the current requirement, but leave the access for new plugins. In the software, there are three aspects needed to be extended in the future:

- Adding new sound effects
- Adding new sound effect parameters
- Interface to other devices, like MIDI

Chapter 3 Software Design

3.1 Design concept

Since the software requirements were specified, the next step was planning a solution that fulfilled all those requirements. The design concept was to identify the key problems from the requirements and provide solutions to each one of them. In order to do so, it is important to look at the test procedure as a whole and to see how the software fits into it. There are three stages to perform a test.

Stage 1: Test preparation
Before the real test is started, the testers need to initialize both the hardware and the software, especially this sound manipulation software. The tester selects the warning sounds and imports them into the software, associates these sounds with traffic events, and also configures the connection between the simulation software and this software.

Stage 2: Test run-up
During the test, the driver operates the driving equipment and hence interacts with the vehicle simulator based on a 3D animation which is projected on a wall. The warning system consists of two speakers and the sound manipulation software. In this stage, the only thing the tester needs to do is to click the ‘Start’ button and then the warning system works automatically. However, a real-time user control is needed so that the tester is able to change settings promptly in the case that something goes wrong. For example, the testers can manipulate the mechanism of the sound effect and also disable and enable traffic events during the test, if needed to test the particular events and warning sounds.

Stage 3: Interview and questionnaires
In this stage, the tester discusses the opinions of the driver concerning the warning sounds by asking the driver a list of questions. Thus, the software at this stage is believed to have fulfilled its purpose. However, after each test, a historical log file should be created by the software that records the detailed information about the test - start and end times of the test, the choice of warning sounds, numbers of crashes, etc.

Therefore, the software should perform the following functions:

- Sound playback
- Sound manipulation
- Receive and sift out environment inputs
- Store and organize the input data
- Act as a decision-making agent for sound activation
- Incorporate a GUI to allow for better settings and control during the test

3.2 Software structure

Based on the concept above, the software is divided into four functional modules - communication module, real-time sound manipulation module, and tracking and trigger module and user interface module. Figure 3.1 shows the 4 modules and the bold arrows indicate data flows between each of the modules.

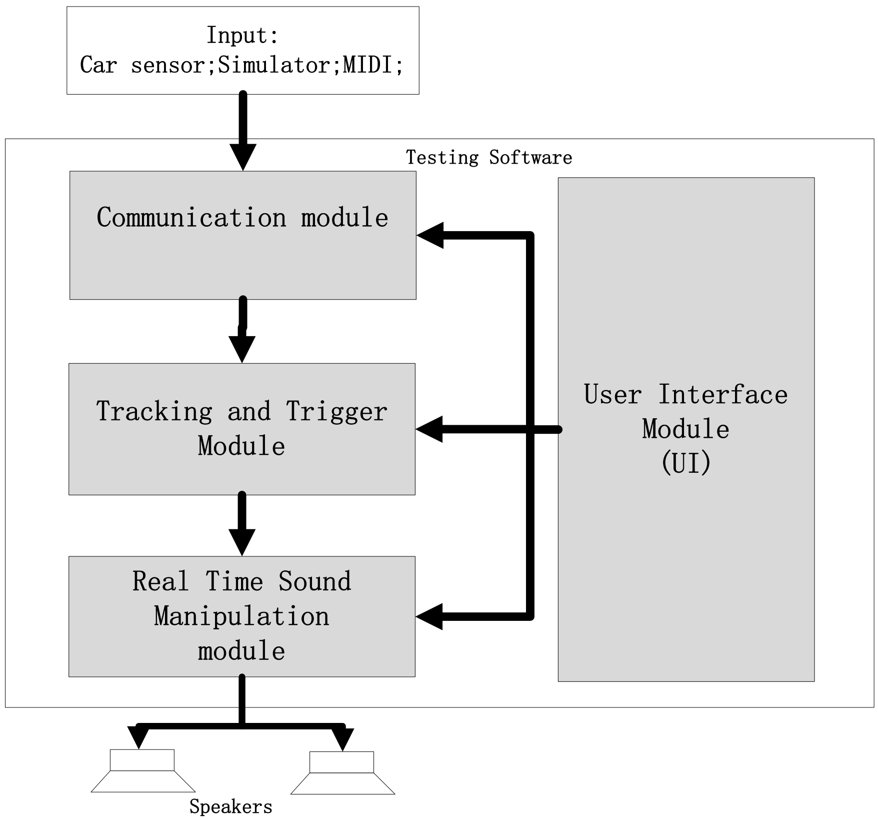


Figure 3.1: Software structure

3.2.1 Real time processing module

This module provides a series of functions concerning the sound playback, real time sound manipulation and sound mixing. It is designed for two different situations depending on the target users:

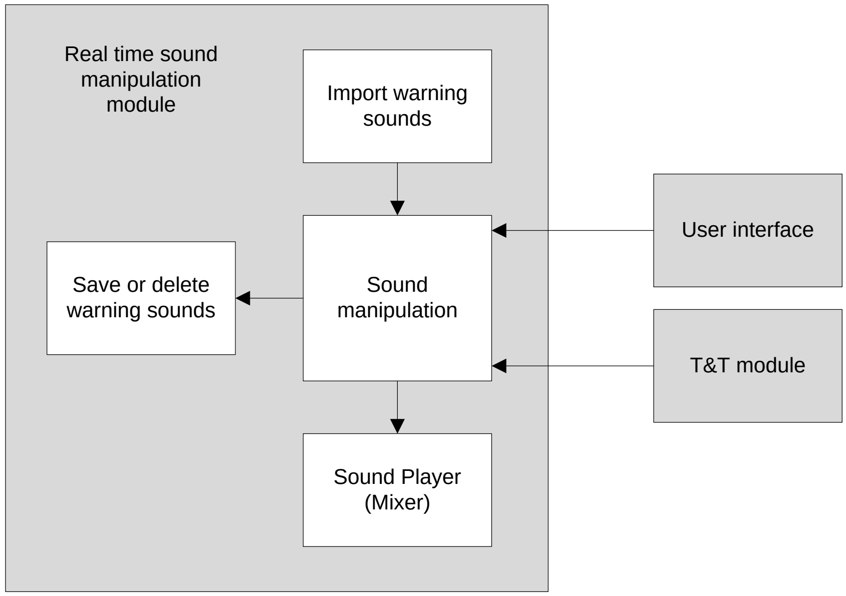


Figure 3.2: Structure of the real-time sound manipulation module

Within this module, three functional units are designed

1) Import, save or delete warning sounds

   It is a unit which deals with the sound files. It imports the sound files from the hard disk into the software, saves the modified warning sounds back to the disk, and deletes unwanted sounds. In additional, it also transforms the sound files into a unique encoding format (stereo sound wave with 44.1 kHz sampling rate), so that the software is able to mix multiple sounds. There are too many encoding formats to choose from like MP3, WAV and so on. In the present thesis, the software adopts the WAV format since it retains more details of the sound than MP3. A relatively higher sampling rate (44.1 kHz) is used. The Two-channel encoding enables the isolated manipulation on each of the channels, thus helping the spatial panning of the sound. Therefore, all the sound files, before being used as warning sounds, are transformed into this format.

2) Sound manipulation

   Sound manipulation is the core of the software, and it can be used under two conditions – during sound editing and during the test. During sound editing, the tester manually modifies sound parameters to create a new sound. Hence the software provides flexibility to the testers, in a way that they can change the pitch or volume by dragging track bars. During the test, the software works automatically with the vehicle simulator. The trigger and tracking module decides which warning sounds should be activated and passed into the sound manipulation unit. The tracking process keeps passing the corresponding data based on the sound effects settings that the tester has set up. The detailed implementation will be shown in the next chapter.

3) Sound player (Mixer)

Similar to the sound manipulation unit, the sound player works in two different stages: while sound editing, the testers can mix the selected sounds together and hear the result. In order to handle multiple sounds, the mixer is designed that has the capacity to mix up to 100 sounds. According to the simulation design, one traffic scenario, at one moment, can have up to 4 traffic objects and each object can have a maximum of 3 traffic events. Moreover, each traffic event is consists of one or up to 10 warning sounds. That meant that a maximum of 40 sounds could be used for playback, at an instant, and hence 100 mixing channels were enough for the driving test.

3.2.2 Communication module:

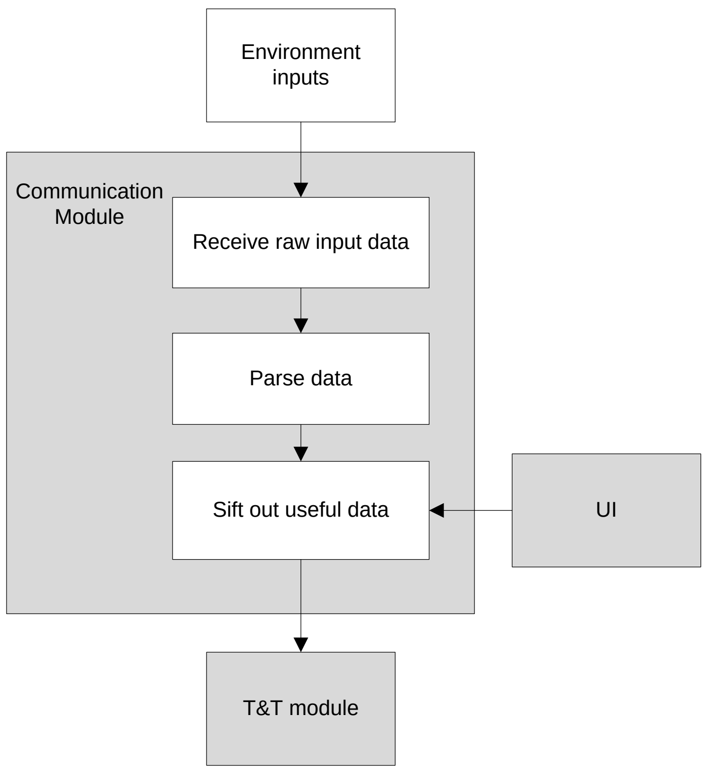


Figure 3.3: Structure of communication module

The vehicle simulator, STISIM, provides a RS232 serial communication port between two computers. This means that an interface is needed to access the serial port and receive the input data. Furthermore, the data type on the serial port is the byte array and needs to be parsed into useable format, the floating-point array. STISIM defines that there are up to 50 types of data for one single object, such as speed, longitude distance, lateral distance, acceleration, TTC (Time to collision), steering wheel angles, acceleration caused by brake, etc. However, not all the received data is used during the test. The software allows the tester to choose usable input parameters to use and to relate them with traffic events.

3.2.3 Tracking and triggering module:

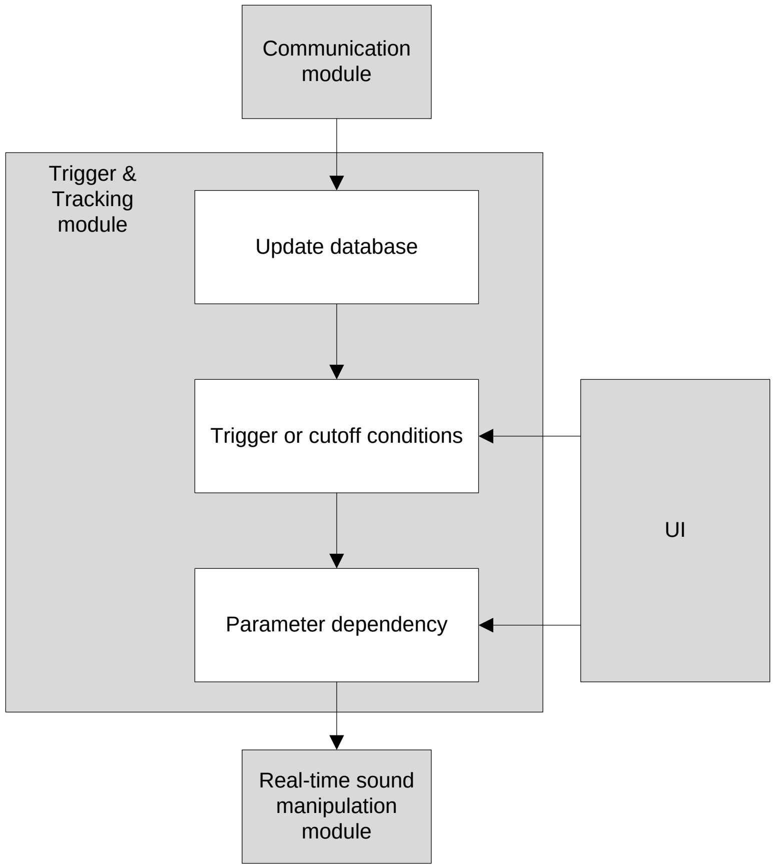


Figure 3.4: Structure of Trigger and tracking module

As figure 3.4 indicates, this module can be seen as a bridge between the communication module and the real time sound manipulation module. It functions every time it receives data stream and calculates the data to determine whether to trigger warning sounds or not.

When the data is available, the first task is to replace the old data with a new one for all the traffic objects, including the driver's vehicle itself and surrounding objects. For the sounds for the traffic events that are inactivated, the next step is to examine if the new data have reached their trigger threshold which were set by the tester. And for the sounds for the traffic events that are activated, it would be to examine if they have reached the cutoff threshold.

Once warning sounds are activated, sound effects work automatically according to the 'Sound Effects' set-ups (see section 3.2.4). T&T module collates the data from the chosen effects, input parameters and weights, and then calculates the variation of each of the chosen effects, which would be utilized by the sound manipulation module later on.

3.2.4 User interface

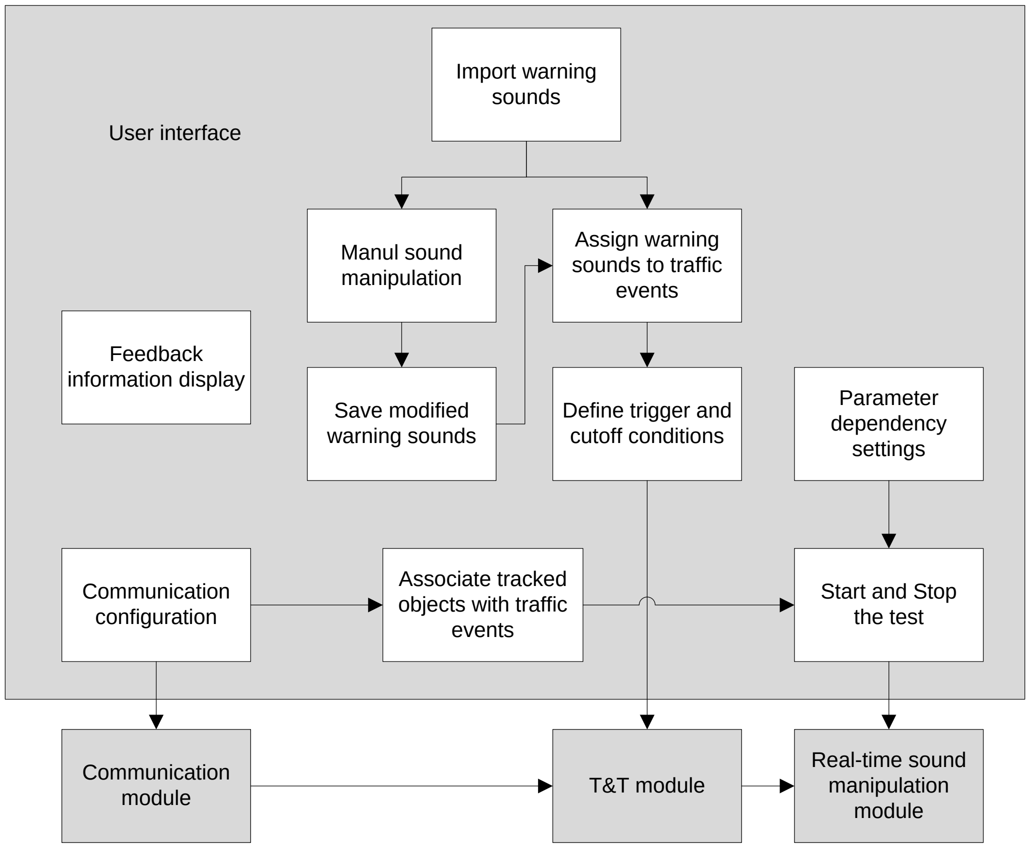


Figure 3.5: Structure of user interface module

The user interface is a module that works directly with the tester, while other three modules work behind stage. The main reason behind the tester using the GUI is to accomplish the testing set-ups, such as sound library, connections among different modules, initializing simulator scenario and so on. The main interface is shown in figure 3.6. It can be divided into five parts which are:

- a window for displaying operation history;
- three buttons to start, stop or exit the test;
- a sound library settings that allows the tester to import and modify traffic events and warning sounds;
- a sound effects settings where the tester set up parameter dependency before and during the test;
- a simulator control that enables the tester to configure the simulator connection, associate objects with the events and observe the data change during the test.

It is not necessary to follow this order but needs to perform all the actions in a given step.

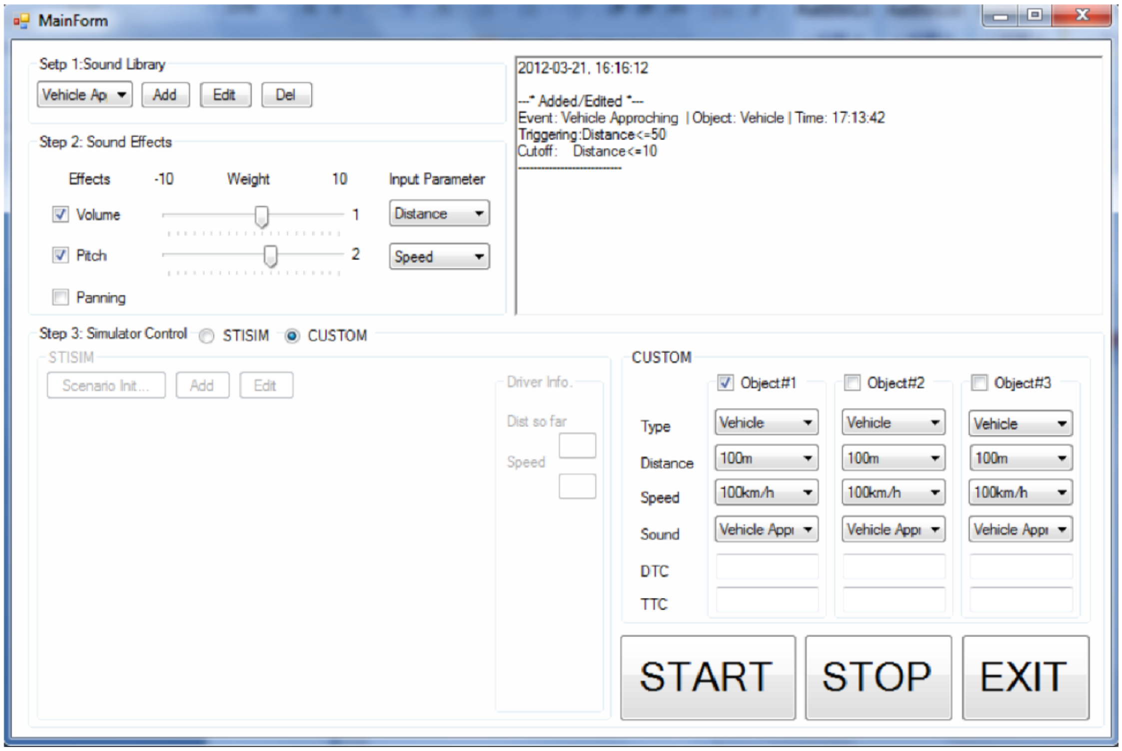


Figure 3.6: Screen shot of the main interface of the software
(an example of 'Vehicle approaching')

3.2.4.1 Sound Library

The sound library has a collection of a set of warning sounds that can be used for different events. By clicking the button '*Add*' or '*Edit*', the figure 3.7 would show up and meanwhile the main interface would be disabled.

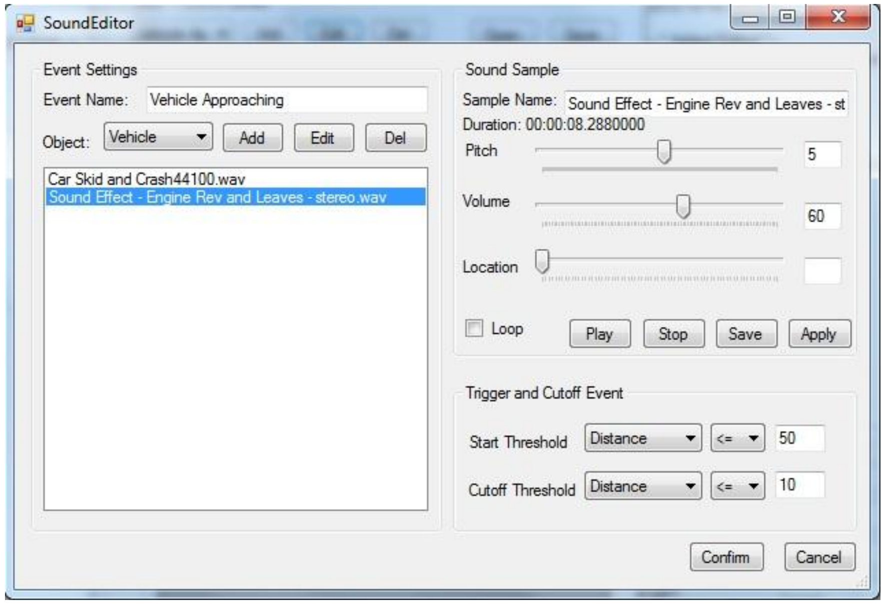


Figure 3.7: Screen shot of Sound Editor

The functions in the interface enable the tester to: (see appendix I for information about GUI components *Combo box*, *List box*, *Text box*, *Check box*)

- Define the name of a traffic event.
- Define the object type of event by choosing from the *Combo Box*.
- Add, save and delete sound samples associated with the traffic event (associate warning sounds with event).
- Select sound samples at the *List Box* to edit or delete.
- Rename the chosen sound sample.
- Play the chosen sound sample.
- Modify the chosen sound sample by dragging the '*Track bar*' marked by Pitch and Volume. To be more accurate, the *Text box* is placed beside of track bar for the tester to enter numbers.
- Save the modified sound sample
- Enable the chosen sound sample to loop when playback, by marking the *Check Box* marked '*Loop*'.
- Set up trigger and cut-off conditions. An example is shown in figure 3.7. It indicates that this traffic event arises when the relative distance is lower than 50 meters and ends when the relative distance is lower than 10 meters.

3.2.4.2 Sound Effects setting

This step allows the tester to define how the sound effects work. The check boxes help to enable or disable sound effects. The 'Weight' track bar indicates the degree of influence of the input parameter on the sound effect. All these components can be real-time manipulated, which allows the tester to change settings during the simulation.

3.2.4.3 Simulator settings

The simulator control has two simulator options. The custom simulator is designed for a pre-test. That enables the testers to hear their sound designs and sound effects before the driver. There are three traffic objects in total and their actions are set to approaching to the driver. The relative distance decreases by the speed that the tester has chosen from combo box. The decreasing rate is 0.1 second which means sound effects changes every time the distance decreases.

Compared to the customized simulator, STISIM simulator is harder to configure owing to its complex scenario structure. One STISIM simulation is made of several scenario blocks; and each block consists of multiple traffic objects which can be vehicle, motorcycle, and pedestrian and so on. The blocks are segmented by the elapsed distance since the beginning of the run. For example, taking 1000 meters as block stride, block 1 starts at 0 meter and ends with 1000 meters, and block 2 starts at 1000 meters and finishes at 2000 meters, and the list goes on.

From the software perspective, the number of scenarios, total length of each scenario and the number of objects within each scenario are required for the test initialization. Unfortunately, none of that information can be found from the serial port that the STISIM provides. Hence, some manual operations are needed. First, the tester creates scenario blocks by opening and editing the 'Scenario Editor' (see figure 3.8). A block is created by enabling the corresponding checkbox and specifying the length of the block by entering the ending length in the text box .

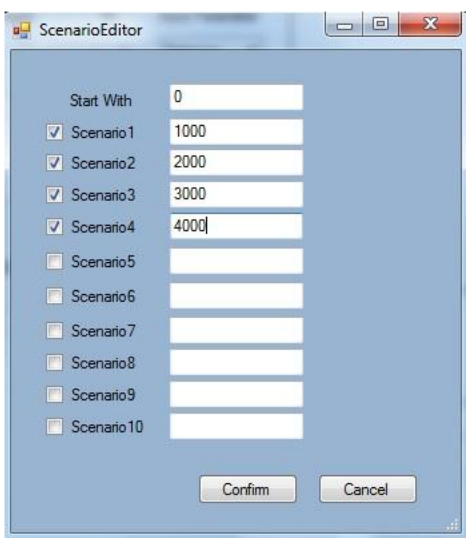


Figure 3.8: Screen shot of STISIM scenario editor

After clicking on the ‘Confirm’ button, the selected number of tabs would appear on the main form (see figure 3.9). A GUI component ‘Tab group’ (see appendix I) is created and each ‘Tab’ indicates one specific scenario block.

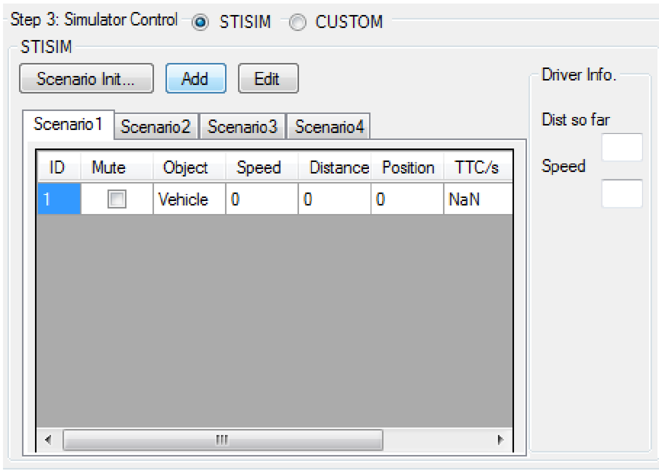


Figure 3.9: STISIM simulator control panel

The next step is to add traffic objects to each block and associate them with the traffic events. As figure 3.10 shows, traffic events would be listed on ‘List Check box’. One object can consist of more than one traffic events.

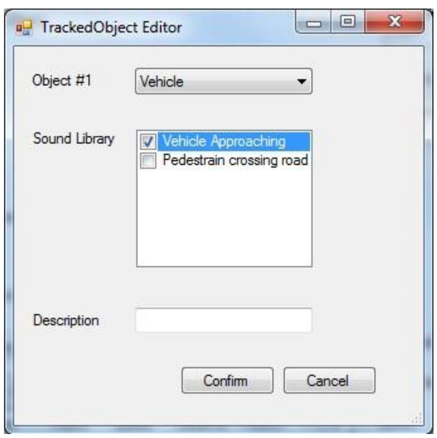


Figure 3.10: Tracked object editor

At run time, the tester is able to change effects and simulator settings. It can disable and enable sound effects, and mute an object sound if necessary.

3.3 Data structure

The data structure addresses the question of storing and organizing data so that it can be used efficiently. In the present thesis, the data structure needs to store the data from the simulator, to assign trigger and cut-off conditions and to associate them with intuitive warning sounds.

At first, an object-based model was designed so that the traffic object was the basic data unit of the software. Each traffic object had its own database to save the STISIM data, trigger and cut-off conditions, and a set of warning sounds. By doing so, when a certain traffic object entered the critical zone, its assigned warning sounds would be produced to inform the driver. However, the flaws of this data type were that the traffic object's behaviors differed over time and it was impossible to use one group of sounds to represent. The testers might want to use different sounds to represent different situation, like the sound of a car tire skidding to indicate the proximity to a leading car, sound a crash to indicate collision. Therefore, an event-based data type was created. This data structure focuses on the particular actions that the traffic object makes.

The design of the data structure corresponds to the structure of STISIM scenario and event-based data model, which can be arranged by following logic:

- STISIM scenario consists of scenario blocks
- One block is made of traffic objects
- Each object might have numbers of events
- Each event is represented by a set of auditory icons
- There is no limit on the number of objects, events and sounds

Therefore, 4 class[1] objects are created and nested together, shown in figure 3.11. The direction of the arrow means that the class object on the right is a subset of the class on the left. The class 'ScenarioBlock' , for instance, includes a group of class object 'TrackedObject'.
Note: This figure only shows data members of a certain class object, not class methods.

1 In object-oriented programming, a class is a construct that is used to create instances of itself – referred to as class instances, class objects, instance objects or simply objects. A class defines constituent members which enable its instances to have state and behavior. Data field members (member variables or instance variables) enable a class instance to maintain state. Other kinds of members, especially methods, enable the behavior of class instances. Classes define the type of their instances. (See http://en.wikipedia.org/wiki/Class_%28computer_programming%29)

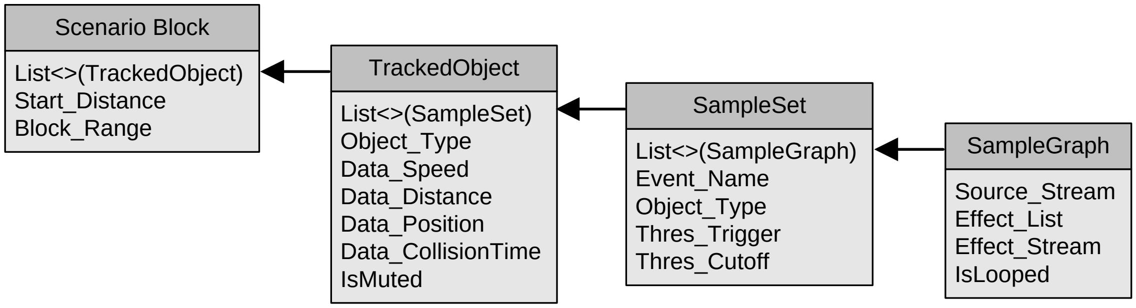


Figure 3.11: Relationship between sound and STISIM driving Scenario

The class '*SampleGraph*' holds the original data source of the warning sound and creates '*Effect_Stream'* data type which combines the original sound with the sound effects. The '*SampleSet*' is the class object that defines the traffic event and its activation conditions. The '*TrackedObject*' corresponds to a certain traffic object and is used for saving the input data.

Chapter 4 Implementation

4.1 Real-time sound manipulation module

The forms of file formats, compression, and data storage of sound are all changing since the past, but the underlying mechanisms for playback and manipulation have not varied much. The digital audio playback unit reads the sound file into the computer memory in the forms of byte array. An example of the data structure of digital sound wave which is encoded in a standard 16-bit 44.1 kHz stereo format is shown in figure 4.1. Sound wave is encoded alternately, which means the bytes of respective channels alternate in a single stream (LRLRLR, etc.). The DAC (Digital Analog Convertor) can recognize the structure of the audio stream and convert them into analog signals and split them into different channels, and then that sound is presented to the listener.

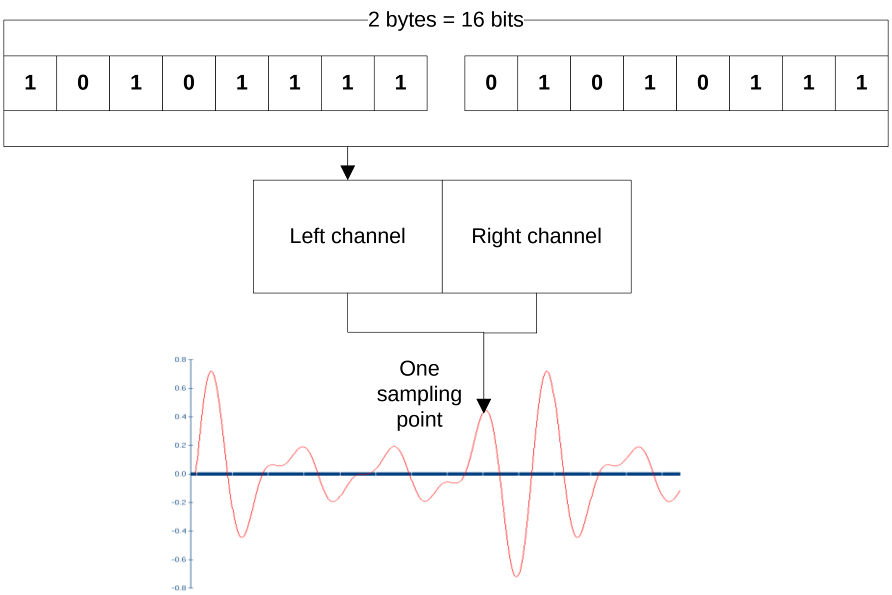


Figure 4.1: The data structure of a single sampling point of sound wave

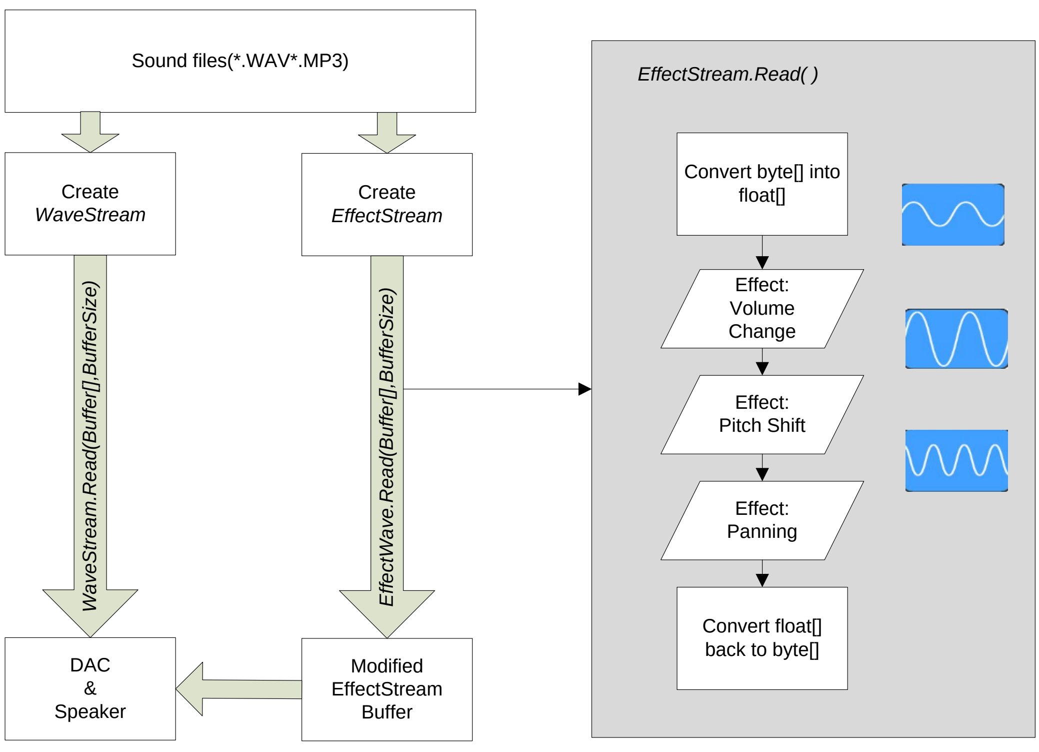
In the present thesis, the C# library *NAduio* provides a collection of functions for digital sound playback [6]. It reads the sound file by creating the *NAduio.WaveStream* class and simply plays this audio stream using the class method [2] - *NAduio.WaveStream.Play( )*. However, it is not that easy for sound manipulation since the binary numbers in which the sound wave is encoded cannot directly be modified and there is no existing function for manipulation. The next section would discuss the implementation of sound manipulation by creating a new class *EffectStream*, an inherited class of *WaveStream*.

[2] Method, in object-oriented programming, is a subroutine (or *procedure*) associated with a class. Methods define the behavior to be exhibited byinstances of the associated class at program run time. (see: http://en.wikipedia.org/wiki/Method_(computer_programming))

4.1.1 Sound manipulation

The method, *NAudio.WaveStream.Read( )*, leaves an interface for the sound effect plug-ins because of the mechanism of the *WaveStream.Play( )*. During the sound playback, the *WaveStream.Play( )* keeps calling the *WaveStream.Read( )* method at a specific interval rate to read a buffer from the sound stream and transfers this buffer into the DAC. The interval is shorter than the duration of the sound buffer, in order to ensure the continuity of the sound.

Therefore, the concept of sound manipulation is to modify the *WaveStream.Read( )* method by embedding the sound effects within this method, so that the sound effects would function every time the method is called. The *EffectStream* Class is the derived class of *WaveStream*, and has its own *Read( )* method. The first task of this new method is to convert the byte array data type into floating point array. And then the sound effects are done by the overridden method called *Sample(ref float spl0, ref float spl1)*, where *spl0* and *spl1* represents the value of the left and right channels respectively.



(a) Mechanism of digital audio playback (b) Flowchat *of EffectStream.Read()*

Figure 4.2: Real-time sound manipulation module

The word ‘overridden’ means that this method allows the subclass to provide a specific method implementation. In other words, the subclasses can implement different functions by coding the same method, while they share the other features from the parent class. It allows the usage of one *Effect* class framework to create different sound effects by modifying the specific methods.

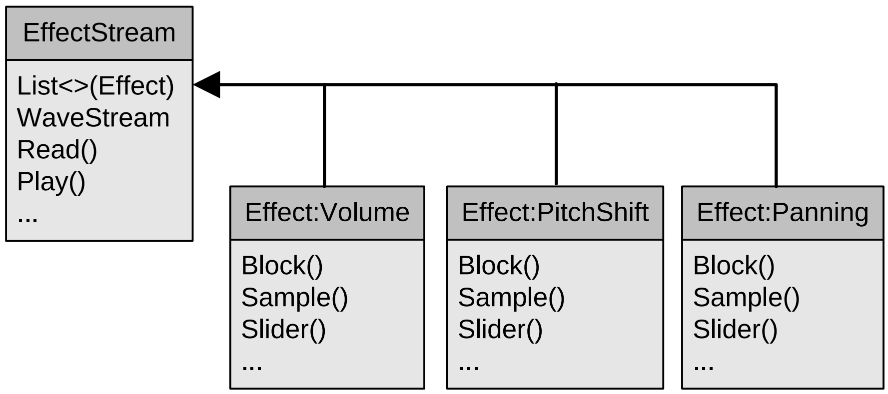


Figure 4.3: The data structure of the class *EffectStream* and *Effect*

The figure 4.3 indicates that the *EffectStream* consists of a list of sound effects, which individually is represented by creating different subclasses of *Effect (Volume, PitchShift, and Panning).* Each effect has its own *Block()* and *Sample()* method. For example, to lower the volume by 50%, the method *Volume.Sample()* could be coded as follows:

```
public override void Sample(ref float spl0, ref float spl1)
{
    spl0 *= 0.5f;
    spl1 *= 0.5f;
}
```

4.1.2 Sound mixing

In order to handle multiple sounds, a player was needed that was capable of mixing all the warning sounds, which could be done by the class - *NAudio.Mixer*. The method *Mixer.Init( )* initializes a certain amount of mixer’s channels (*Mixer.Wavechannel* class) and each of the channels holds an *EffectStream*. Adding a new *EffectStream* to the mixer or removing one from the mixer is done by using the methods *Mixer.AddInputStream()* and *Mixer.RemoveInputStream()*. However, the problem was that these two methods restricted the use of the mixer. The length of the mixer is fixed when adding the first sound stream, and the mixer would stop working when it reached the end of the stream. In simpler words, the sounds could not be played back when the first sound stream ends. Therefore, the *AddInputStream( )* and *RemoveInputStream( )* methods were unsuitable for the sound activation.

An alternative method was found by using class members of *WaveChannel* - *Volume* and *Position*. *WaveChannel.Volume* allows the mixer to modify the volume of each of the mixer channels individually. *WaveChannel.Position* indicates the current playback status of each of the mixer channels. For example, if *WaveChannel.Volume* is set to 0, the corresponding *WaveChannel* would be silence. And if *WaveChannel.Position* is set to 0, it means the mixer will play the certain channel from the first stream buffer.

The principle of the sound activation is simply to unmute and mute warning sounds in an audio mixer. As the tester has set up the traffic events completely and start the test, all the warning sounds are initially mixed together and set to silence. By doing so, each of the sounds is assigned an address of mixer channels. Once the assigned sound is activated for a traffic event, the corresponding mixer channel is unmated. In opposite, the cut-off function is to mute related sounds again. An example below shows that the 10th sound stream in *WaveChannel* is activated.

WaveChannel[10].Volume = 1;
WaveChannel[10].Position = 0;

To ensure that the length of the mixer was sufficient enough to support the simulation test, a silence *BaseStream* with a long duration was created so that the mixer worked until the test finishes. Normally, the duration of one simulation scenario ranges from several minutes to half hour, depending on the performance of the driver who operates the vehicle simulator. If the driver drives slower than the average speed that the tester has designed, the duration would last longer than expected time but no more than 30 minutes owing to the time limitation of the scenario design used in the previous test. Therefore, the length of *BaseStream* is set up to 1 hour.

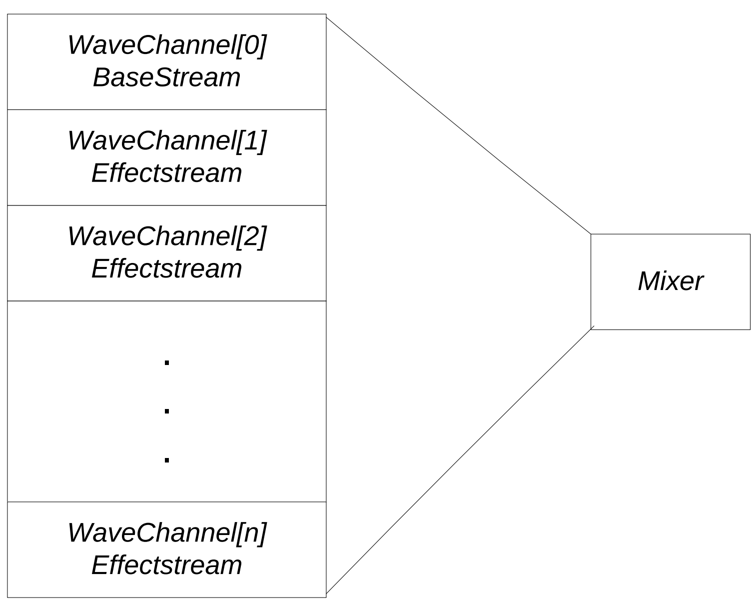


Figure 4.4: Channel structure of mixing player

4.2 Communication module - Working with STISIM

4.2.1 Data Structure on the serial port

The STISIM Drive [7] is a fully interactive, PC-based driving simulator with unlimited customization potential, and is ideal for a wide range of research and development applications concerning the driver, the vehicle, and the environment (road, traffic, pedestrians, visibility, etc.). It provides a maximum of 50 types of variables for the user to select for communication. During the simulation run, it continuously organizes the traffic information into a byte array and sends this array to the other computer through a RS232 serial port. The format of the data stream is relatively simple with a 4 byte header [255 255 255 255] and followed by the user-selected data. Each variable that is selected for output is passed as a 4 byte single precision value.

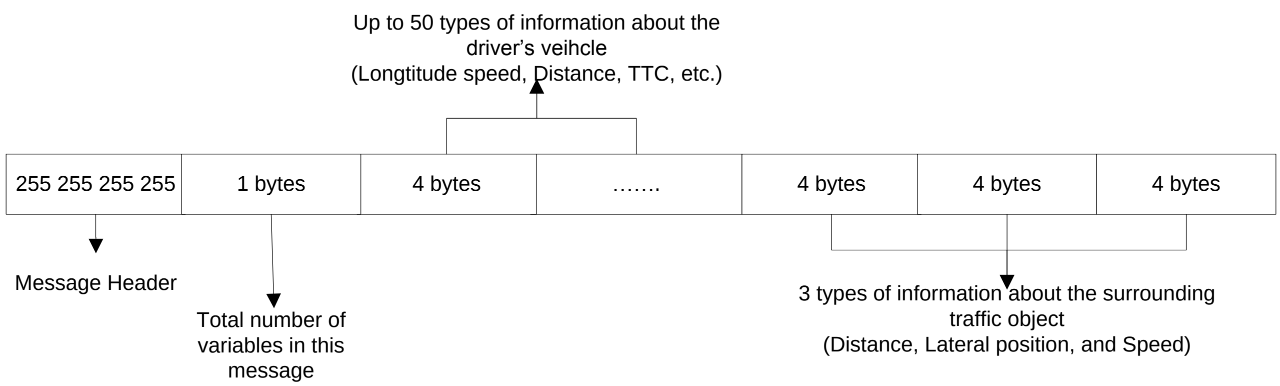


Figure 4.5 Structure of data stream receive from the STISIM

Once the simulation test is started, there is always a data stream available on the serial port. However, its validity of those data is unknown. Thus finding the valid data stream is the first and the most important step. The following equation defines the calculation of *Frame_Size*, the length of a valid data array. Whenever needed, the communication module reads 2*Frame_Size of byte array from serial port to ensure that the received data stream contains a complete set of valid data and then finds the address of the message header.

$$Frame\_Size = 5 + 4*nr\_of\_selected\_varibales + 4*3*nr\_of\_surrounding\_objects \qquad (1)$$

After finding the head of the valid data, the *data parse* method is called to transform the raw data type from byte array into floating point values, by using the built-in function – *bitconvertor.ToSingle().*

4.2.2 STISIM scenario switch

As mentioned in section 3.2.4, a simulation run consists of several scenarios, and each of the scenarios might have the different variable selections and the different numbers of traffic objects. It means that the software needs to adjust the settings for different scenarios in order to avoid data collisions caused by switching the scenario. The figure 4.6 shows a flowchart to solve this problem.

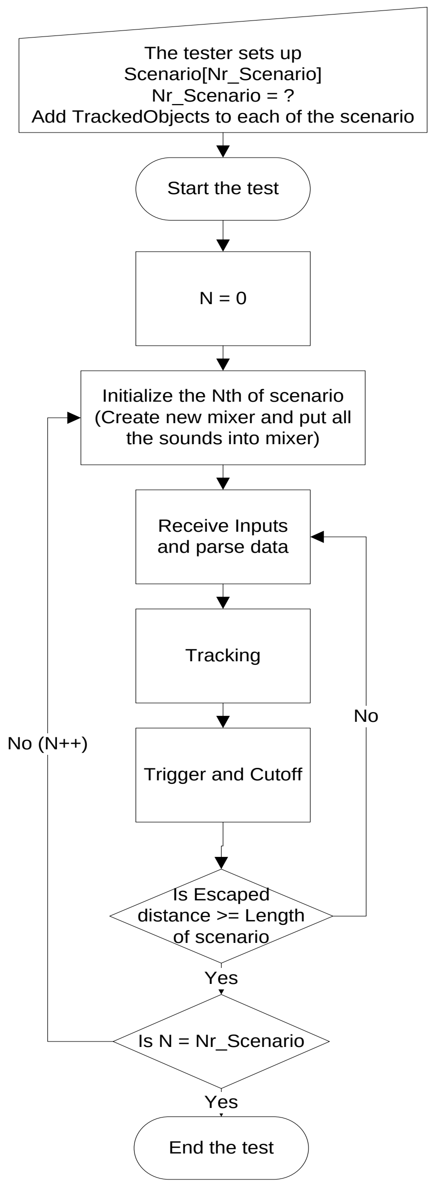


Figure 4.6: The flowchart of scenario switching

The key point of this solution is the initialization of a new scenario. Once the escaped distance has exceeded the length of the current scenario, the software would release the currently used scenario, mixer and tracked objects, and then start the next scenario. Initialization of a scenario is a process that combines the complete set of sound samples of all the tracked objects into the mixer, and assigns the value *0* to each of *WaveChannel.Volume*.

4.3 Tracking and trigger module

4.3.1 Tracking process

Tracking is a process of updating information of all the tracked objects in a given scenario that includes the driver's vehicle and the surrounding traffic objects. The tracking processing does not only transfers the parsed data into corresponding objects, but also calculates several types of variables that reflect the relationship between a certain surrounding object and the driver. An example is shown in figure 4.7, where only one traffic object exists in scenario besides the driver. The value of *Object1.TTC* with respect to the *Driver* could not be acquired from parsed data array, but could be calculated by knowing the value of the *Distance* and *Speed* of both *Driver* and *Object1*. Similar to TTC, the value of relative speed, distance, acceleration and so on, could be acquired by the same means.

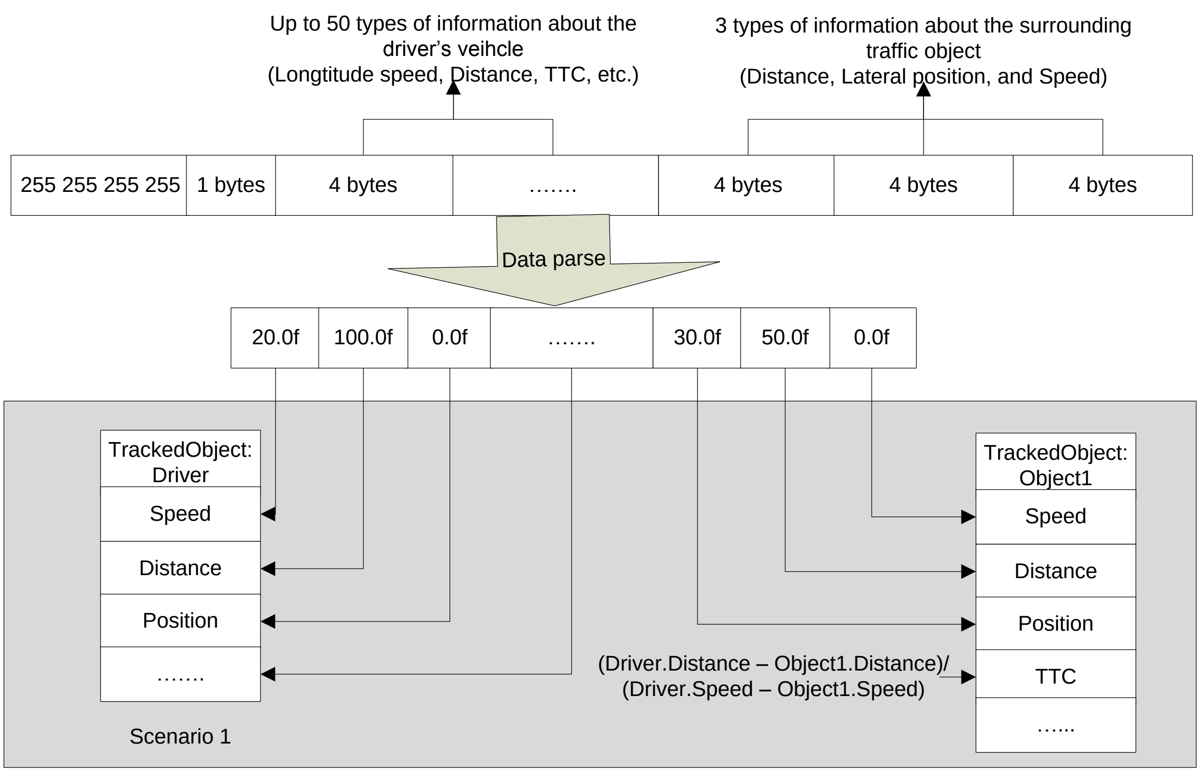


Figure 4.7: An example of tracking process

4.3.2 Trigger process

One tracked object can have several sets of sound samples associated with that particular event. The triggering process checks triggering conditions for each sound sample set. Figure 4.8 shows the complete flow chart for the triggering processing.

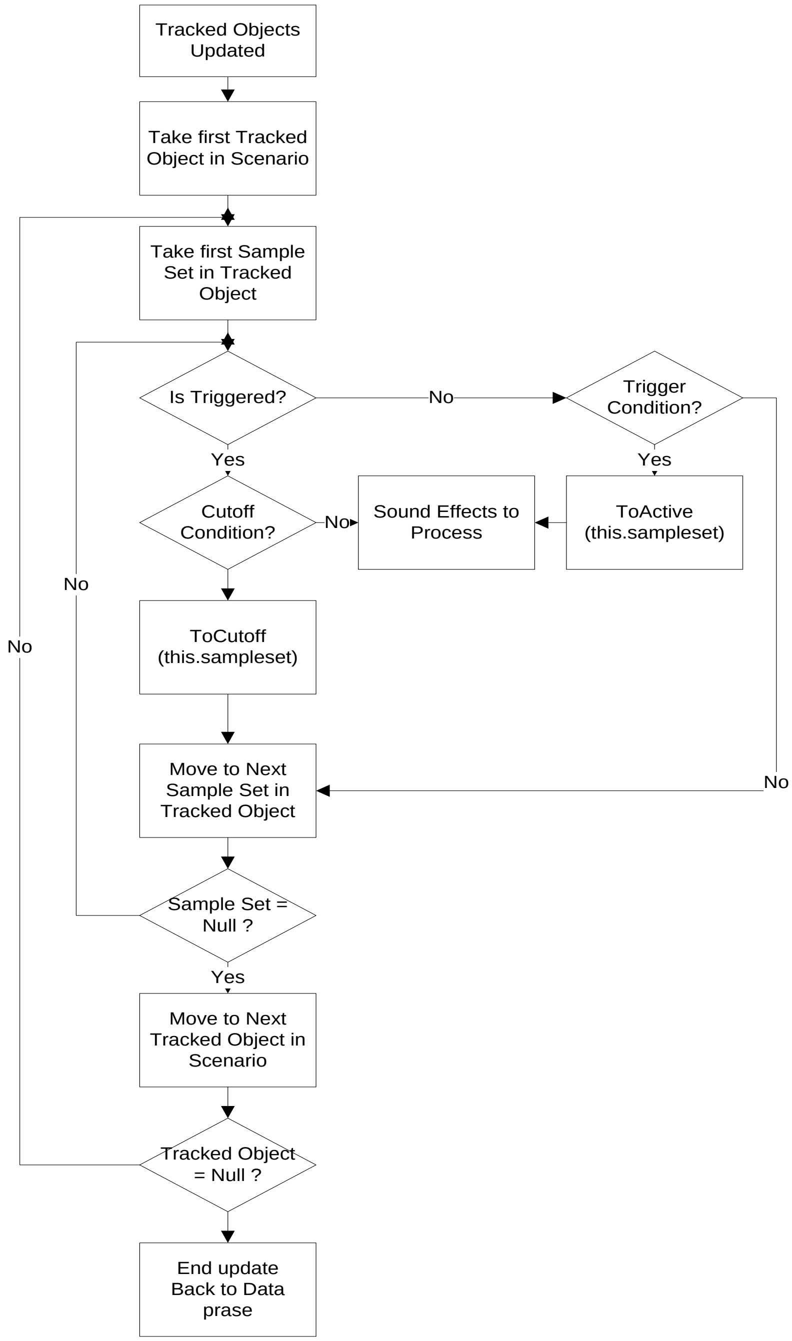


Figure 4.8: Flowchart of trigger process

To simply the sound activation, the trigger and cutoff conditions simply depend on one of input parameters - Distance, Speed, Time to Collision, and Position. Two Boolean variables, *IsTriggered* and *IsCutoffed*, are designed as the flags to prevent from redundant activation. As long as the warning sound is triggered, the value of *IsTriggered* is set to *True*. It means that the trigger process will not examine the trigger condition for this sound track but will examine only the cutoff condition, so that one warning sound of certain traffic event is activated only once.

Chapter 5 Verification and Future Development

5.1 Verification

Several tests were conducted to evaluate the performance of the software by answering the following questions:

1. Is there any latency when manipulating the sound?
2. Does the sound activation work? And is there any delay when the trigger event is called?
3. Are the sound effects considered to be intuitive by the driver?
4. How many sound channels can the mixer handle?
5. How many objects can the software track?

The answers are mentioned below:

There were no delays with the sound activation and manipulation. However, the quality of the sound effects was relatively poorer than expected. The Volume Change effect indeed gave the driver the feelings of approaching and departing of the objects. The Pitch Shift effect worked perfectly but hardly created the Doppler Effect. The hearing experience was better with binaural reproduction using headphones. Therefore, an advanced algorithm for the spatial panning is needed in the future development.

The intuitiveness of the warning sounds is subject to the driver's own experience. Some of them can easily tell the meanings of each warning sound, while others do not. However, it is the purpose of the software that not to give a best solution but to test different types of sounds and sound effects and to improve the auditory icon designs over time.

To test the tolerance of the software, Up to 20 objects were used in one scenario block, and each of the objects associated with 3 traffic events, where one event was represented by 3 warning sounds. Therefore, the heaviest workload for the mixer was 180 channels in one test run. It turned out that the software worked really well, even though it was annoying to have so many warning sounds around. Normally, the tester would design a scenario with 2 to 4 objects so that the performance of the driver related to different objects would be easily distinguished at the final data analysis. Thus the software is capable enough to handle multiple objects and warning sounds.

5.2 Discussion and future Work

The testing software provides a foundation for HMI researcher and sound designer to work with the vehicle simulator – STISIM and to evaluate the in-vehicle information systems. Even though it can satisfy the needs of the current testing scenario, it can be improved in following aspects.

1) Better sound activation
   Trigger event must be improved in the future. For now, the software adopts a very simple trigger rule, which cannot satisfy all the traffic situations. The next step is to make the software smarter. It can determine whether the situation is urgent or not, set different priority to the traffic events. If several events happen at the same time, it should trigger the most dangerous one. It means that the software should have a decision making unit, where the trigger and cutoff conditions are not only related to one parameter, but combination of parameters based on the understanding of the current traffic situation. It is very complicated mission but necessary.

2) Better sound effects
   The sound effects are the essence to make the sound intuitive. The *Panning* effect, in the present thesis, needs to be improved to work with the loud speakers. And also the *Pitch Shift* effect could be improved by implementing a more advanced algorithm.

   Another improvement could be increasing the parameter dependency of the sound effects. The volume effect now has one parameter dependency which is the relative distance. But, in the future, there might be using the combination of distance and speed for this effect. Of course, the last improvement can be made by adding new sound effects.

3) Better user interface
   In the present thesis, leaving flexibility to the user has made the software so many set-ups that the user spends too much time on the test settings rather than running the test. In some cases, configuring STISIM scenario, each object will be associated with sound event. In one test run, there will be 5 to 6 scenarios and up to 20 objects within each scenario, which means that the same operation repeats over 100 times to complete only one function - adding new objects. Not to forget that if the tester makes mistakes when running a simulation, they are going to re-do the set-ups all over again. To solve this problem, the software must optimize the setting steps or provide a default setting, so that the tester does not need to do the setting every time when opening the software.

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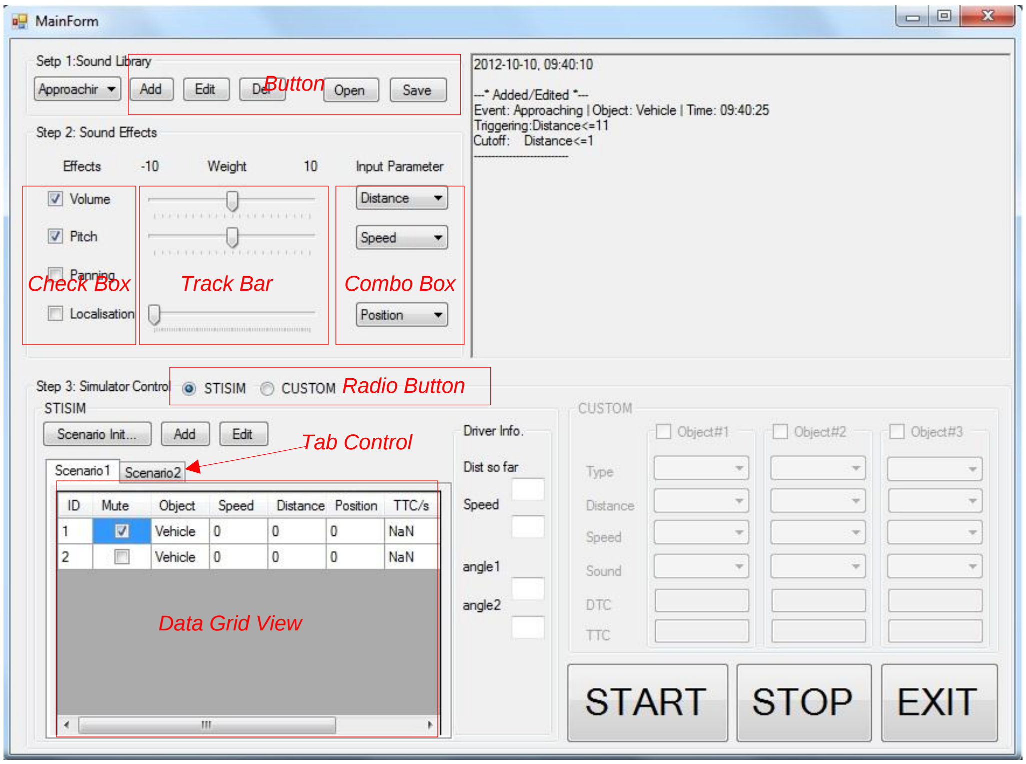
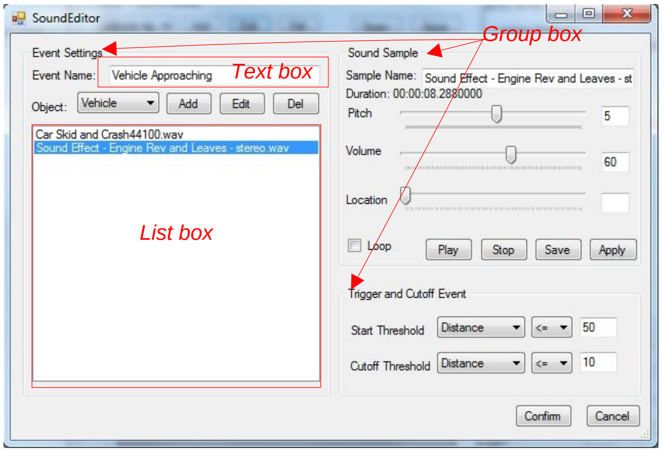
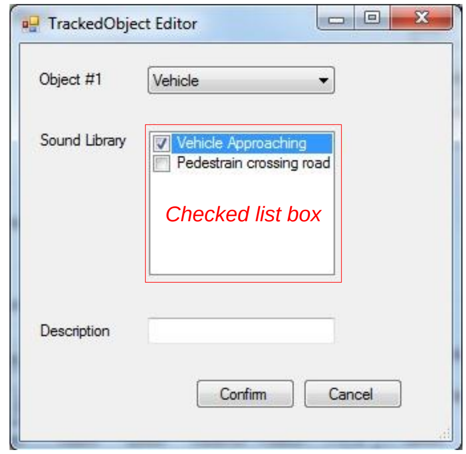
Appendix I GUI controls in C#



1. Button: Raise an event when the user clicks it
2. Check Box: Enables the user to select or clear the associated option
3. Track Bar: enables the user to choose between a range of value by sliding a small bar along on another bar
4. Combo Box: is a drop-down list of items from which the user can make a selection either by clicking an item in the list or by typing into a box.
5. Radio Button: enables the user to select a single option from a group of choices when paired with other radio buttons.
6. Tab control: manages and displays to the user a related collection of tabs that can contain controls and components.
7. Data Grid View: displays rows and columns of data in a grid that the user can customize.





8. Textbox: is a control in which the user inputs data from the keyboard. This area also can display information.
9. List box: is a control in which a list of items is displayed. The user can make a selection from the list by clicking on any item. Multiple elements can be selected.
10. Checked list box: displays a list of items with a check box on the left side of each item so that the user can choose or clear the associated option.
11. Group box: displays a frame around a group of controls with an optional capture.

Appendix II Codes

```
Effectstream Read method:

public override int Read(byte[] buffer, int offset, int count)
        {
            int read;
            lock (sourceLock)
            {
                read = source.Read(buffer, offset, count);
            }
            if (WaveFormat.BitsPerSample == 16)
            {
                lock (effectLock)
                {
                    Process16Bit(buffer, offset, read);
                }
            }
            else if (WaveFormat.BitsPerSample == 32)
            {
                //Process32Bit(buffer, offset, read);
            }

            return read;

        }
```

Effectstream method for sample level manipulation

```
private void Process16Bit(byte[] buffer, int offset, int count)
        {
            int samples = count / 2;
            foreach (Effect effect in effects)
            {
                if (effect.Enabled)
                {
                    effect.Block(samples);
                }
            }

            for (int sample = 0; sample < samples; sample++)
            {
                // get the sample(s)
                int x = offset + sample * 2;
                short sample16Left = BitConverter.ToInt16(buffer, x);
                short sample16Right = sample16Left;
                if (WaveFormat.Channels == 2)
                {
                    sample16Right = BitConverter.ToInt16(buffer, x + 2);
                    sample++;
                }

                // run these samples through the effect
                float sample64Left = sample16Left / 32768.0f;
                float sample64Right = sample16Right / 32768.0f;
                foreach (Effect effect in effects)
                {
                    if (effect.Enabled)
                    {
                        effect.Sample(ref sample64Left, ref sample64Right);
                    }
                }

                //Debug.Assert(Math.Abs(sample64Left) <= 1.0);
                //Debug.Assert(Math.Abs(sample64Right) <= 1.0);

                sample16Left = (short)(sample64Left * 32768.0f);
                sample16Right = (short)(sample64Right * 32768.0f);

                // put them back
                buffer[x] = (byte)(sample16Left & 0xFF);
                buffer[x + 1] = (byte)((sample16Left >> 8) & 0xFF);

                /*byte[] b = BitConverter.GetBytes(sample16Left);
                Debug.Assert(b[0] == buffer[x], "DOH");
                Debug.Assert(b[1] == buffer[x+1], "DUH");*/

                if (WaveFormat.Channels == 2)
                {
                    buffer[x + 2] = (byte)(sample16Right & 0xFF);
                    buffer[x + 3] = (byte)((sample16Right >> 8) & 0xFF);
                }
            }
        }
```

```
#region triggering event corresponding to a sound set

        public string EventName { get; set; }
        public string ObjectName { get; set; }
        public int Index { get; set; }
        public int channelIndex{ get; set;}

        public float StartVar { get; set; }
        public int StartIndex { get; set; }
        public int StartOpt { get; set; }
        public float StartThres { get; set; }

        public float CutoffVar { get; set; }
        public int CutoffIndex { get; set; }
        public int CutoffOpt { get; set; }
        public float CutoffThres { get; set; }

        public bool IsTriggered { get; set; }
        public bool IsCutoffed { get; set; }


        public bool ToTrigger
        {

            get
            {
                switch (StartOpt)
                {
                    case 0:
                        return StartVar <= StartThres;
                    case 1:
                        return StartVar >= StartThres;
                    case 2:
                        return StartVar == StartThres;
                    default:
                        return false;
                }
            }
        }
        public bool ToCutoff
        {

            get
            {
                switch (CutoffOpt)
                {
                    case 0:
                        return CutoffVar <= CutoffThres;
                    case 1:
                        return CutoffVar >= CutoffThres;
                    case 2:                           ;
                        return CutoffVar == CutoffThres;
                    default:
                        return true;
                }
            }
        }
```

Audio Mixer: Initialize Mixer, to active a sound set, to cut off a sound set

```
public void Init(SimulatorScenario scenario)
        {
            foreach (TrackedObject trackedObject in scenario.TrackedObjectList)
            {
                foreach (SampleSet samples in trackedObject.SelectedSoundTracks)
                {
                    samples.channelIndex = channelNo;
                    foreach (SampleGraph sample in samples)
                    {
                        channels[channelNo] = new WaveChannel32(new
WaveOffsetStream(new EffectStream(sample.Effects, sample.SourceStream)));
                        Mixer.AddInputStream(channels[channelNo]);
                        channels[channelNo].Volume = 0;
                        channelNo++;
                    }

                }
            }
        }

        public void ToActive(SampleSet samples)
        {
            int i = samples.channelIndex;
            foreach (SampleGraph sample in samples)
            {
                channels[i].Volume = 1.0f;
                channels[i].Position = 0;
                sample.Rewind();
                i++;
            }

        }

        public void ToLoop(SampleSet samples, SampleGraph sample)
        {
            int i = samples.channelIndex + samples.IndexOf(sample);
            if (channels[i].Position >= channels[i].Length)
                channels[i].Position = 0;


        }

        public void ToCutoff(SampleSet samples)
        {
            int i = samples.channelIndex;
            foreach (SampleGraph sample in samples)
            {
                channels[i].Volume = 0;
                i++;
            }
        }
```

STISIM data parse and tracking code

```
private void port_DataReceived(object sender, SerialDataReceivedEventArgs e)
        {
            // read two size of byte buffer from port data stream
            if (port.BytesToRead >= 2*frame_size)
            {
                byte[] bytes = new byte[2 * frame_size];
                port.Read(bytes, 0, 2 * frame_size);
                port_DataParse(bytes);
            }
        }

        private void port_DataParse(byte[] bytes)
        {
            int offset = 0;
            int stride = 4;
            bool protector = false;
            //List<float[]> input = new List<float[]>();
            // find message header [255,255,255,255]
            for (int i = 0; i < 2 * frame_size; i++)
            {
                if (bytes[i] == 255 && bytes[i + 1] == 255 && bytes[i + 2] == 255
&& bytes[i + 3] == 255)
                {
                    offset = i;
                    protector = true;
                    break;
                }
            }

            //useful data followed by header, now transfer byte data stream to
float array
            if (protector)
            {
                int cnt = bytes[offset + stride];
                float[] floats = new float[cnt];
                for (int j = 0; j < cnt; j++)
                {
                    floats[j] = (float)Math.Round(BitConverter.ToSingle(bytes,
offset + (j + 1) * stride + 1),2);
                }
                driver.Speed = floats[0];
                driver.Distance = floats[1];
                int k = 0;
                protector = false;
            }
        }
```