Faculty of Mathematics and Information Science Warsaw University of Technology

Biometric fusion system for human recognition using face and voice

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History of changes

Date	Author	Description	Version
14.10.2017	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	Initial version	1.0
19.10.2017	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	Chapter *Glossary* added	1.1
15.11.2017	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	Chapter *Technical project* added	1.2
25.11.2017	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	Chapter *Thesis division into projects* added	1.3
05.12.2017	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	MDC added	1.3.1
03.01.2018	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	Chapters *Tests results* and *User guide* added	1.4
07.01.2018	Kuśmierczyk Aleksander Sławińska Martyna Żaba Kornel	Chapter *Tests results* updated	1.4.1

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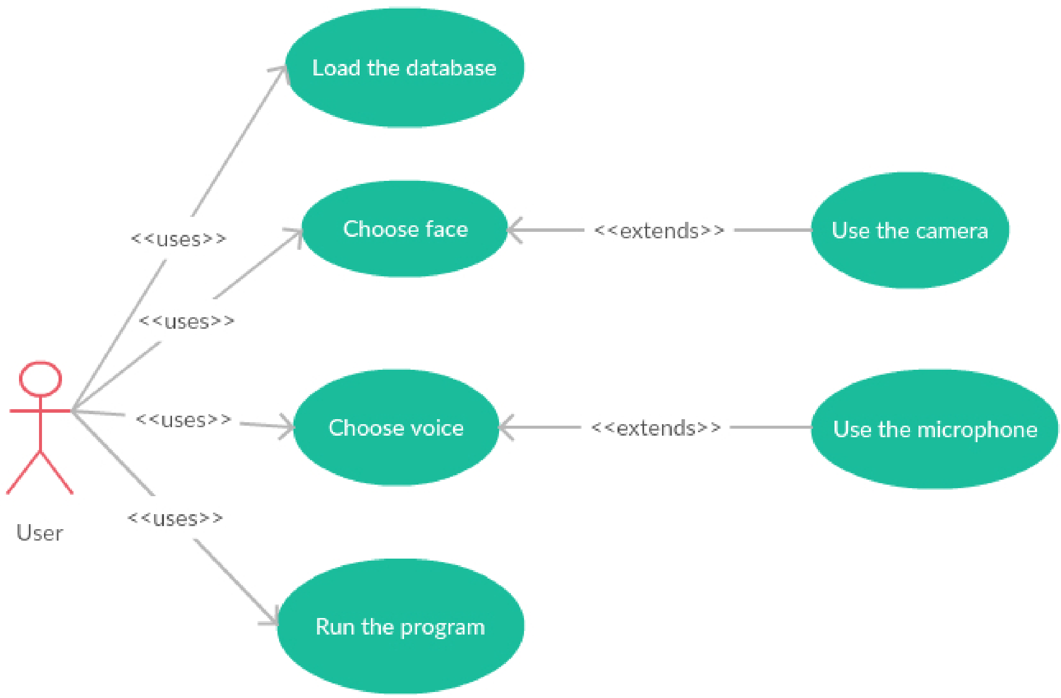
1 Specification

1.1 Executive summary

This system is designed for identification and/or verification of human. With the use of biometric algorithms based on extracting face and voice features, the program recognizes the user with the sufficient degree of compliance. The acquisition of the aforementioned data is done by the portable or built-in camera with a microphone.

1.2 Functional requirements

1. The system should display graphical user interface with options.
2. The system should allow the user to choose what to identify/verify first: face or voice.
3. The system should identify and/or verify the face:
   3.1. The system should use the camera for capturing the image of the face.
   3.2. The system should compare PV’s biometric with the biometrics stored in the database.
4. The system should identify and/or verify the speech/voice:
   4.1. The system should use the microphone for recording the speech.
   4.2. The user should say the specific word in order for the system to identify his/her voice.
   4.3. The system should compare PV’s biometric with the biometrics stored in the database.
5. After collecting PV’s face and speech data, the system should compare collected results with the database and give an answer whether the PV already belongs to the database collection (verifying stage).



Actor	Name	Description	System response
User	Load the database	Load the database for machine learning purposes	The system loads the database from location pointed by the user.
	Choose face	Choose face for identification	The program starts the camera
	Choose voice	Choose voice for identification	The program starts the microphone.
	Run the program	Run the program which compares the PV's data with the database collection	The system returns the result of recognition.
	Use the camera	Use the camera for capturing the face of PV	The system captures the face of PV with camera.
	Use the microphone	Use the microphone for recording the speech of PV	The system records the voice of PV with microphone.

User stories

Attempt of PV verification:

The user starts the program, chooses which biometric should be used in recognition process as first. Then the user uses the appropriate hardware to capture the required input or uses the previously prepared data for the program to use in order to recognize the PV. The user reads the results of the verification.

1.3 Non-functional requirements

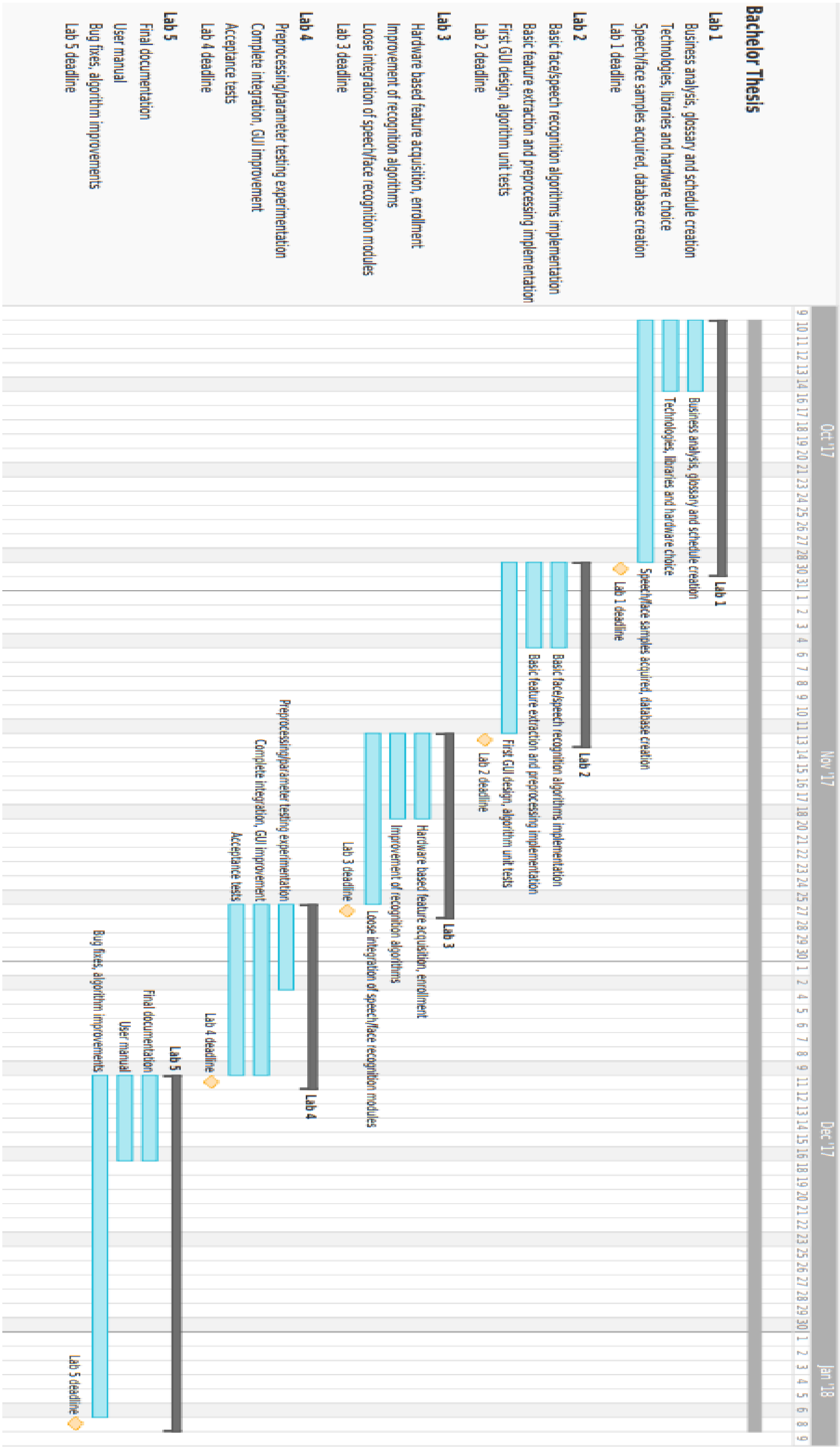
1. The system does not have any login options.
2. The system allows identifying one face/voice at the time.
3. The system should support the Windows operating system.
4. The system should work in an offline mode.
5. The system response time depends on the size of the database.
6. The system's hardware connects by USB cable: camera and microphone, or is built-in the workstation.

Below is the table that describes the URPS requirements of the program.

Requirement	No of requirement	Description
Usability	1	The software is user friendly, as the only requirement for the use of the program is to load the database and to capture the image or to record the speech.
Reliability	2	The system is a desktop application, after manual start-up it is ready to work.
	3	The program handles the incorrect data types in the database, and shuts down if a hardware malfunction occurs.
Performance	4	The system returns the verification result with the delay basing on the database size.
	5	The program verifies the PV using two biometric features, thus achieves the sufficiently good verification correctness.
Supportability	6	In order to use the application, the user is required to have Windows operating system.
	7	The program requires a built-in or a USB-connected camera with a microphone.
	8	In order for the program to function correctly, it requires a database consisting of a large enough samples of face images and voice recordings.

1.4 Project schedule

DEADLINE	MILESTONE	LAB DATE
2017-10-14	Business analysis, glossary and schedule creation	LAB 1 30.10.2017: Business analysis, Glossary, Supplementary spec, Schedule
2017-10-14	Technologies, libraries and hardware choice	
2017-10-28	Speech/face samples acquired, database creation	
2017-11-04	Basic face/speech recognition algorithms implementation	LAB 2 13.11.2017: Technical project, Choice of development model
2017-11-04	Basic feature extraction and pre-processing implementation	
2017-11-11	First GUI design, algorithm unit tests	
2017-11-18	Hardware based feature acquisition, enrolment	LAB 3 27.11.2017: Source code of non-integrated modules, unit tests
2017-11-18	Improvement of recognition algorithms	
2017-11-25	Loose Integration of speech/face recognition modules.	
2017-12-02	Pre-processing/parameter testing experimentation	LAB 4 11.12.2017: Integrated modules, smoke test, acceptance tests
2017-12-09	Complete integration, GUI improvement	
2017-12-09	Acceptance tests	
2017-12-16	Final documentation	LAB 8.01.2018: Working v.1.0, user manual, code/docs, acceptance tests results, list of changes
2017-12-16	User manual	
2018-01-06	Bug fixing, algorithm improvements	
2018-01-06	Others	


Bachelor Thesis
Lab 1
Business analysis, glossary and schedule creation
Technologies, libraries and hardware choice
Speech/face samples acquired, database creation
Lab 1 deadline
Lab 2
Basic face/speech recognition algorithms implementation
Basic feature extraction and preprocessing implementation
First GUI design, algorithm unit tests
Lab 2 deadline
Lab 3
Hardware based feature acquisition, enrollment
Improvement of recognition algorithms
Loose integration of speech/face recognition modules
Lab 3 deadline
Lab 4
Preprocessing/parameter testing experimentation
Complete integration, GUI improvement
Acceptance tests
Lab 4 deadline
Lab 5
Final documentation
User manual
Bug fixes, algorithm improvements
Lab 5 deadline
Oct '17
Nov '17
Dec '17
Jan '18

1.5 Glossary

Biometric – feature owned by human, which allows verification of an individual

CDF - Cumulative Distribution Function

DCT **-** Discrete Cosine Transform

DTW **-** Dynamic Time Warping

Enrolment – process of acquisition of data of the new PV

Face features – face biometric

Feature acquisition – process of collecting data

Feature extraction – process of extracting biometric information from data

FT **-** Fourier Transform

Identification – the automatic identification of living individuals by using their physiological and behavioral characteristics

MDC – Minimum Distance Classifiers

MFCC - Mel Frequency Cepstral Coefficients

Offline mode – the mode of program, which does not require internet connection

PV – person being verified

Recognition - refers to the automated recognition of individuals based on their biological and behavioral traits

Verification - an identity authentication process used to confirm a claimed identity through uniquely identifiable biological traits

Voice features – voice biometric

2 Technical project

2.1 Selected technology

The technology we have chosen for our project is the **C# .NET 4.6** due to the familiarity with this framework of our group. Furthermore, the availability of the online sources of information for this technology allowed us to search for the solutions of encountered implementation problems in the web. Throughout the work it has proven to be stable and provided useful functionalities.

For the development and management of the database used in this project we have chosen **MSSQL Server 2012**. This platform has proven to be stable and using the 2012 version allowed us to find multiple solutions to the problems we have encountered during the work over the project. Besides, our group is familiar to its structure due to the previous experience with this program.

For capturing of images and the recording of sound we have chosen to use external program - **FFmpeg** as it provided us with easy to adapt commands for the aforementioned activities. As hardware solution we have chosen Logitech HD Webcam C270 as it provides the necessary video and audio data required by the project.

2.2 Development model

The goal of our project is to provide the method of identification and/or verification of a person, the study over this subject is still being developed, thus the model we had to adapt for its creation had to be flexible and allow us for frequent theoretical discussion. The development technique we have chosen is the **iterative waterfall model** with the elements of extreme programming. Such choice is based on the need of flexibility for implementation and theoretical discussion, which this model provides, as well as the constant assessment of the chosen methods and their adaptation. All the methods implemented are tested afterwards and the results are discussed with the supervisor in order to gain more understanding over them and to establish further direction for the project. Elements of extreme programming we have adapted to our model were the periodic replacement of the roles of people in the team. Such decision is made in order to allow all the members to follow and to understand the theoretical reasons behind the decisions over the methods of human recognition. Furthermore, this supports better quality of the code produced and allowed for frequent discussion over the problems encountered during the development process.

2.3 System architecture

System architecture is composed of layered modules, each representing different responsibility.

List of modules:

Common - module responsible for database access, as well as containing utility classes used by multiple of below modules.

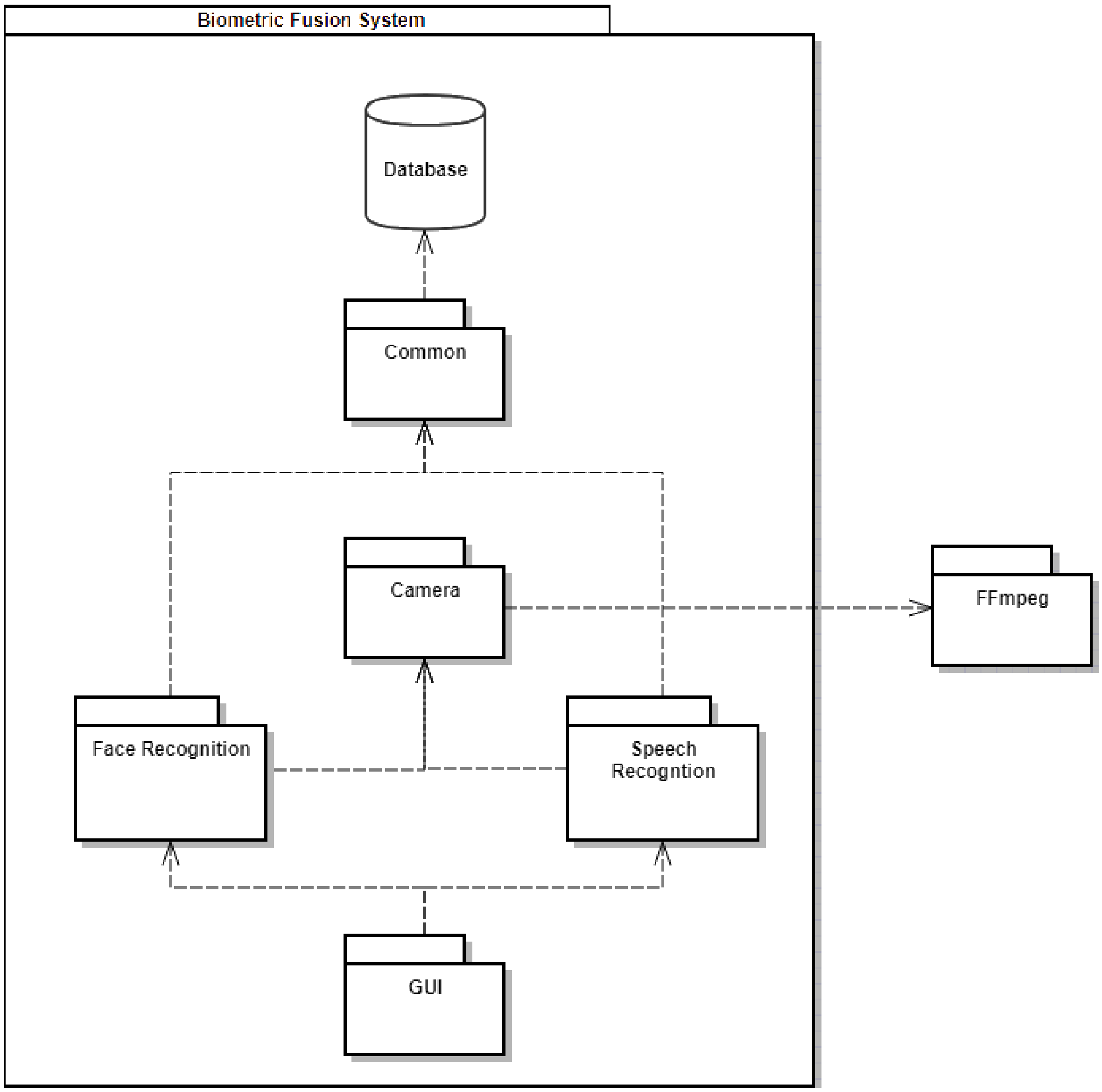
Camera - module responsible for acquisition of the face images and speech recordings. Depends on the scriptable, external video and audio recording program **FFmpeg**.

FaceRecognition - module containing algorithms for face image preprocessing, feature extraction and recognition.

SpeechRecognition - module containing algorithms for speech recording preprocessing, feature extraction and recognition.

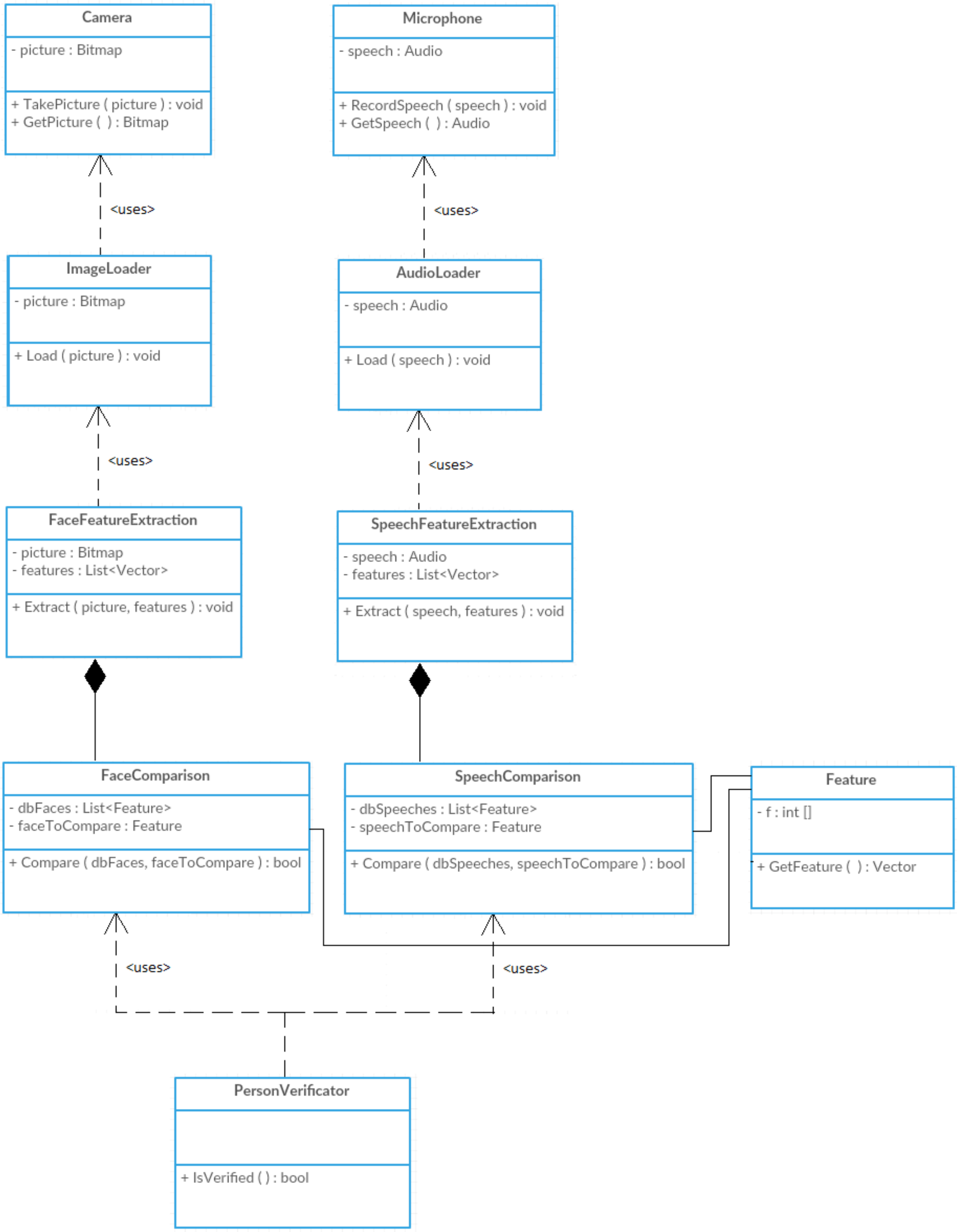
Gui - module defining user interface and interaction. Depends on the .NET desktop application framework **Windows Presentation Foundation.**

Package diagram of system modules:



2.4 Class diagram

The below picture represents the class diagram of the whole system.



3 Thesis division into projects

3.1 General division into projects

Classes description of *Camera* project:

- **WavFile** - class containing the structure of wav. files: Header, Left channel and Right channel
- **WavHeader** - class containing the information of the wav. file stored in the header of the file.
- **WavReader** - class which handles reading of the speech input data.

Classes description of *Common* project:

- **DbConnection** - class containing the information necessary for establishing connection with the database: Server, Data Source, Initial Catalog, User ID and Password to the corresponding user.
- **FaceRepository** - class responsible for the storing and recovering face biometric data from the database.
- **SpeechRepository** - class responsible for the storing and recovering of speech biometric data from the database.
- **Person** - class containing the information of the person whose data is to be used for the verification/identification process.
- **IVerifier** - Interface defining the function which verifies the input against the data stored in the database.
- **FourierTransform** - class which applies the ***FT*** onto the samples using the implementation in the MathNet package.

Classes description of *FaceRecognition* project:

- **INormalization** - interface that stores definition of a function Normalize which takes a bitmap as an argument and normalizes it.
- **Histogram** - class that contains a function GetHistogram which returns the three channel color histogram of a bitmap.
- **HistogramEqualization** - class that implements the interface INormalization, the function Normalize is implemented such that it returns the bitmap after histogram equalization (bitmap with greater contrast).
- **GrayscaleConverter** - class that implements INormalization interface, the function Normalize converts the image into the grayscale image.
- **GaborFilter** - class with functions for applying Gabor filter to the bitmap which is an edge detection filter.
- **HistogramFeatureExtraction** - class for creating the face feature vectors basing on the color histograms and calculating moments from those histograms.
- **GaborFilterMagnitudes** - another class for creating the face feature vectors basing on the Gabor filter and calculating mean, standard deviation and skewness using Gabor filter functions.
- **FaceFeatureExtractor** - class that utilizes **HistogramFeatureExtraction** and **GaborFilterMagnitudes**, and combines the final face feature vector from feature vectors calculated in those two classes.
- **MinimumDistanceClassifier** - class used for the verification of the input against the database.

Classes description of *SpeechRecognition* project:

- **Frame** - class containing partial list of samples from given *.wav* file.
- **FrameMaker** - class dividing stream of samples from given .wav file into frames of equal length given length of frame and interval between frames in ms.
- **IWindow** - interface defining function applying a window function to a given frame.
- **GaussWindow** - class applying Gauss window to a given frame, implements **IWindow** interface.

- **HammingWindow** - class applying Hamming window to a given frame, implements **IWindow** interface.
- **Periodogram** - class calculating power spectrum coefficients of a given frame.
- **MelConverter** - class converting given frequency in Hertz to Mel scale and back.
- **MelFilterbank** - class generating filters for given range of frequencies and sample rate and computing a set of ***MFCC*** for generated set of filters and spectral density coefficients of given frame.
- **SpeechFeatureExtractor** - class extracting a feature vector from a given list of frames. Utilizes a **IWindow** implementation, **Periodogram** and **MelFilterbank** to generate a combined feature vector of ***MFCC*** of each frame.
- **DynamicTimeWarping** - class used for the verification of the input against the database using the ***DTW*** algorithm with a given threshold.

Classes description of *UnitTests* project:

- **DynamicTimeWarpingTest** - set of tests checking correctness of ***DTW***.
- **FrameMakerTest** - set of tests checking correctness of frame generation from the audio recording.
- **GaborFilterTest** - set of tests checking correctness of the Gabor filter algorithm.
- **HistogramTest** - set of tests checking correctness of histogram creation.
- **MelConverterTest** - set of tests checking correctness of conversions between Mel and Hertz.
- **MelFilterbankTest** - set of tests checking correctness of ***MFCC*** generation algorithm.
- **PeriodogramTest** - set of tests checking correctness of periodogram coefficients calculation.
- **WindowFunctionTest** - set of tests checking correctness of calculation of Hamming window and Gauss window.

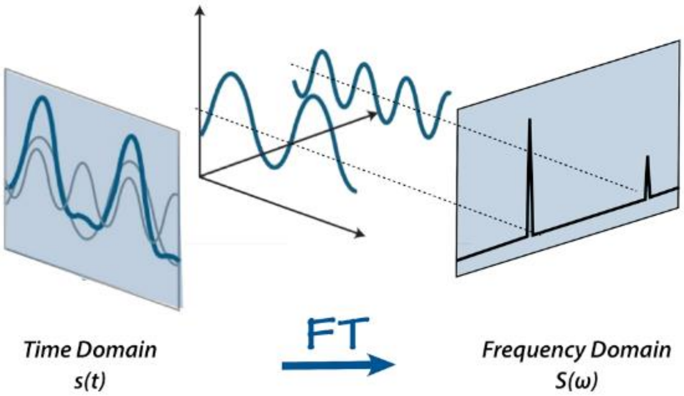
Classes description of *AlgorithmTests* project:

- **SpeechAlgorithmTest** - set of tests checking for successful identification of an input .wav file against set of template .wav files. Each file consists of a recording of a single word '*kurczak*'.
  Files: "aleks_k.wav", "aleks_k2.wav", "aleks_k3.wav",
  "aleks_k4.wav", "kornel_k.wav", "martyna_k.wav"
  Each test checks if different combination of "aleks" template and input recording will be correctly verified.
- **FaceAlgorithmTest** - set of tests checking for successful identification of an input .bmp file against set of template .bmp files. Each file contains a picture of a face.
  We have two pictures of person1 (input and template), one picture of person2 and one picture of person3. We compare the similarities between person1 (input) and
  - person1 (template),
  - person2 picture,
  - person3 picture.

3.2 Classes description in Common project

FourierTransform

Fourier Transform is mapping the signal from time domain to spectral domain.

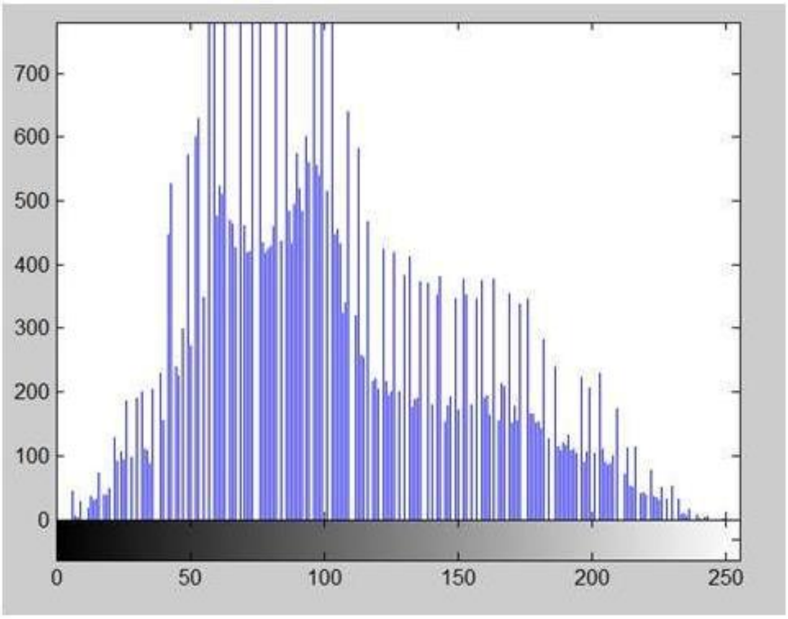


The formula for Fourier Transform calculation is the following:

$$x_k = \sum_{n=0}^{N-1} x_n \cdot e^{-i2\pi kn/N} = \sum_{n=0}^{N-1} x_n \cdot [cos(2\pi kn/N) - i \cdot sin(2\pi kn/N)]$$

3.3 Classes description in FaceRecognition project

Histogram
For every channel, distribution of intensities is calculated.
Sample grayscale image histogram:



HistogramEqualization
Histogram equalization calculates the ***CDF*** for the color channel values, which is also the image's accumulated normalized histogram and applies to the picture. Such operation increases the global contrast for the image, as the intensities of the color level are better distributed on the histogram.
CDF is calculated with the following formula:

$$cdf_x(i) = \sum_{j=0}^{i} p_x(j)$$

where $p_x(i)$ is the probability of an occurrence of a pixel of level i in the image and is calculated with the following formula:

$$p_x(i) = p(x = i) = \frac{n_i}{n}, 0 \le i < L$$

where L is the number of color levels in the channel, n is the total number of pixels in the image and n_i is the number of occurrences of the color level i.
In order to apply this method back to the original picture the final function is used:

$$y = \lfloor cdf(i) * ((max\{x\} - min\{x\}) * min\{x\}) \rfloor$$

where y is the new pixel color level for the channel x from the original color level i

The result of using HistogramEqualization implemented in the project:

before after

GrayscaleConverter
In order to transform the image into grayscale image, the function Normalize loops through every pixel of the image and sets the value for each channel of given pixel to the average value of values of all channels of that pixel.

The pseudocode is the following (actions done for each pixel):
channelValue = (pixel.Red + pixel.Green + pixel.Blue) / 3
pixel.Red = channelValue
pixel.Green = channelValue
pixel.Blue = channelValue

GaborFilter
Gabor filter is an edge detection filter. The edges of an image are detected from different orientations, and next they are placed on top of one another to get the final image with detected edges.

The formula used to calculate Gabor filter is the following:

$$G_{\sigma_y,\sigma_x,f_i,\theta_k}(x,y) = e^{-[\frac{x_{\theta_k}^2}{\sigma_x^2}+\frac{y_{\theta_k}^2}{\sigma_y^2}]} \cdot cos(2\pi f_{ix\theta_k} + \phi)$$

where

$$x_{\theta_k} = x \cdot cos\theta_k + y \cdot sin\theta_k$$

$$y_{\theta_k} = y \cdot cos\theta_k - x \cdot sin\theta_k$$

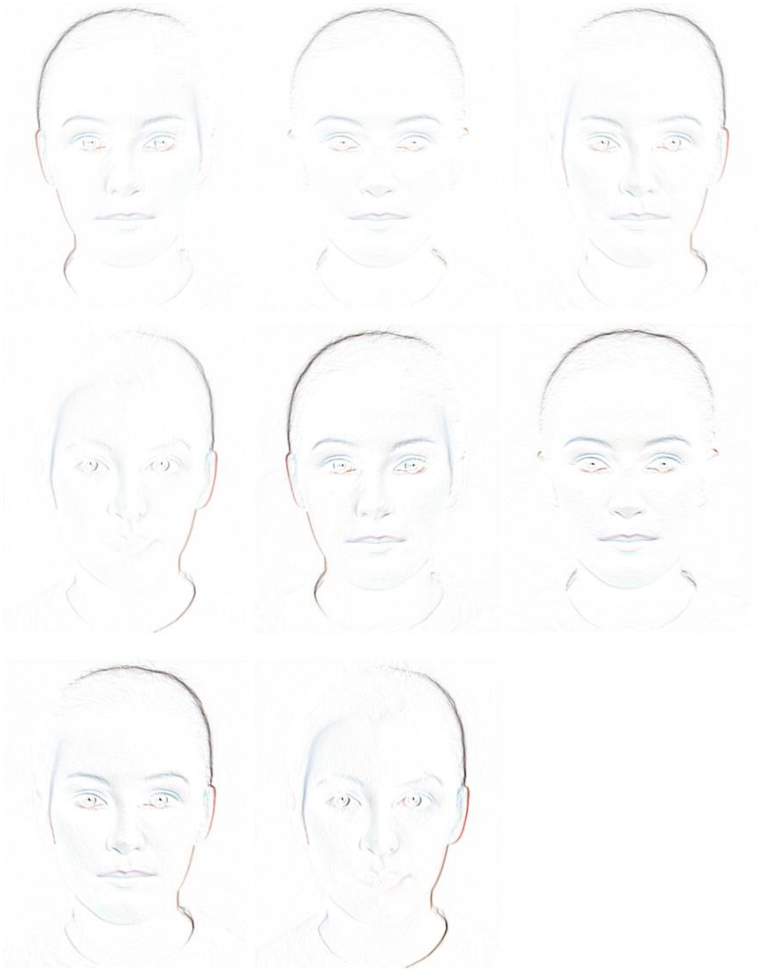
$$\phi = \pi/2$$

$$\theta_k = \frac{k\pi}{n}$$

The next step is to calculate the kernel by applying the formula to particular elements of the kernel:

$$kernel[x,y] = e^{-[\frac{x_{\theta_k}^2}{\sigma_x^2}+\frac{y_{\theta_k}^2}{\sigma_y^2}]} \cdot cos(2 \cdot \pi \cdot \frac{x_{\theta_k}}{waveLength} + \phi)$$

Finally, applying the kernel to the image produces the following images:



After placing them one on top of another, we obtain final image after Gabor filtering:

HistogramFeatureExtraction
Color moment generating formula used in HistogramFeatureExtraction:

$$M_k^1 = \frac{1}{XY}\sum_{x=1}^{X}\sum_{y=1}^{Y} f_k(x,y) \quad \text{for the first order}$$

$$M_k^h = \left(\frac{1}{XY}\sum_{x=1}^{X}\sum_{y=1}^{Y}(f_k(x,y) - M_k^1)^h\right)^{\frac{1}{h}} \quad \text{for any subsequent order defined by h}$$

k - represents the k-th color component.
h - order of the moment
$f_k(x, y)$ - the color value of the k-th color component in the pixel at the coordinates of (x,y)
The moments generated by the aforementioned functions are: mean for the first order, variance for the second order and skewness for the third order, all efficiently describe the color distributions of images[http://shonen.naun.org/multimedia/NAUN/bio/bio-2.pdf]

GaborFilterMagnitudes
This class calculates the Gabor filter magnitudes which are used as a part of face feature vector.

The magnitudes (mean, standard deviation, skewness) are calculated from the following formulas:

$$\mu = \frac{1}{XY}\cdot\sum_{x=1}^{X}\sum_{y=1}^{Y} imageAfterGaborFiltering(x,y)$$

$$std = \sqrt{\frac{1}{XY}\cdot\sum_{x=1}^{X}\sum_{y=1}^{Y} ||imageAfterGaborFiltering(x,y)| - \mu|^2}$$

$$skew = \frac{1}{XY}\cdot\sum_{x=1}^{X}\sum_{y=1}^{Y}\left(\frac{imageAfterGaborFiltering(x,y) - \mu}{std}\right)^3$$

FaceFeatureExtractor
It combines the feature vectors obtained from the classes HistogramFeatureExtraction and GaborFilterMagnitudes into one face feature vector.

Minimum Distance Classifiers
This method analyses the numerical face feature vectors and calculates the Euclidean distance between them in order to further use the calculated distance in determining how similar the given vectors are.

The formula for calculating the Euclidean distance:

$$\mathrm{d}(\mathbf{q},\mathbf{p}) = \sqrt{(q_1 - p_1)^2 + (q_2 - p_2)^2 + \cdots + (q_n - p_n)^2}$$

3.4 Classes description in SpeechRecognition project

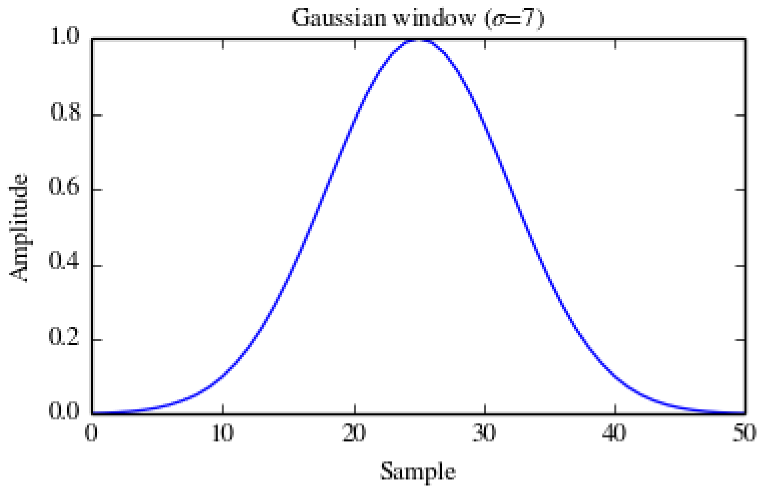
FrameMaker
FrameMaker divides a stream of sampled voice recording into frames of equal length.
Length of frame in samples is obtained by the following formula:

$$F_{len} = t_{len}\cdot S_r t_(len) \quad \text{= frame length in seconds } S_r \text{ = sample rate of recording}$$

Interval between frames is obtained by the following formula:

$F_{int} = t_{int} \cdot S_r t_{(} int)$ = frame interval in seconds S_r = sample rate of recording

GaussWindow
Gauss Window goes over each frame of the audio recording, amplifies the middle of the frame and weakens the beginning and the end of the frame.



HammingWindow
Hamming Window goes over each frame of the audio recording, amplifies the middle of the frame and weakens the beginning and end of the frame.

The formulas used:

$$w(n) = \alpha - \beta \cdot cos(\frac{2\pi n}{N-1})$$

n=sample in a frame N=number of samples in a frame

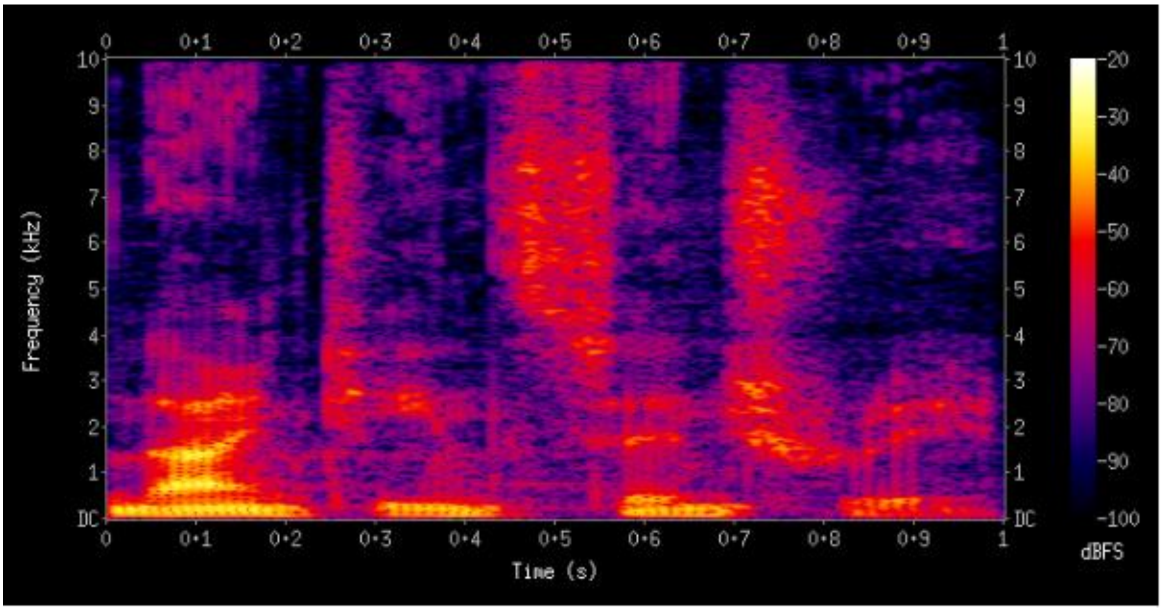
α = 0.53836

β = 0.46164

Periodogram
Periodogram applies ***FFT*** to a *n* long Frame, mapping the signal from time to spectral domain, resulting in *n* complex coefficients. Afterwards spectral density estimates, which describe strength of signal given the frequency, are calculated from first half of ***FFT*** coefficients.
Spectral density estimation of a single coefficient is given by the following formula:

$P(c_i) = \frac{|c_i|^2}{N}$ c_i = *i*-th coefficient of ***FFT*** N = amount of samples in a frame

Below a spectrogram, x axis corresponds to time, y axis to frequency and color spectrum to amplitude.



MelConverter
Mel Converter converts the Mel units into Hertz, and Hertz into Mel.
Formula for converting Mel into Hertz:

$$M^{-1}(m) = 700 \cdot (e^{m/1125} - 1)$$

Formula for converting Hertz into Mel:

$$M(f) = 1125 \cdot ln(1 + \frac{f}{700})$$

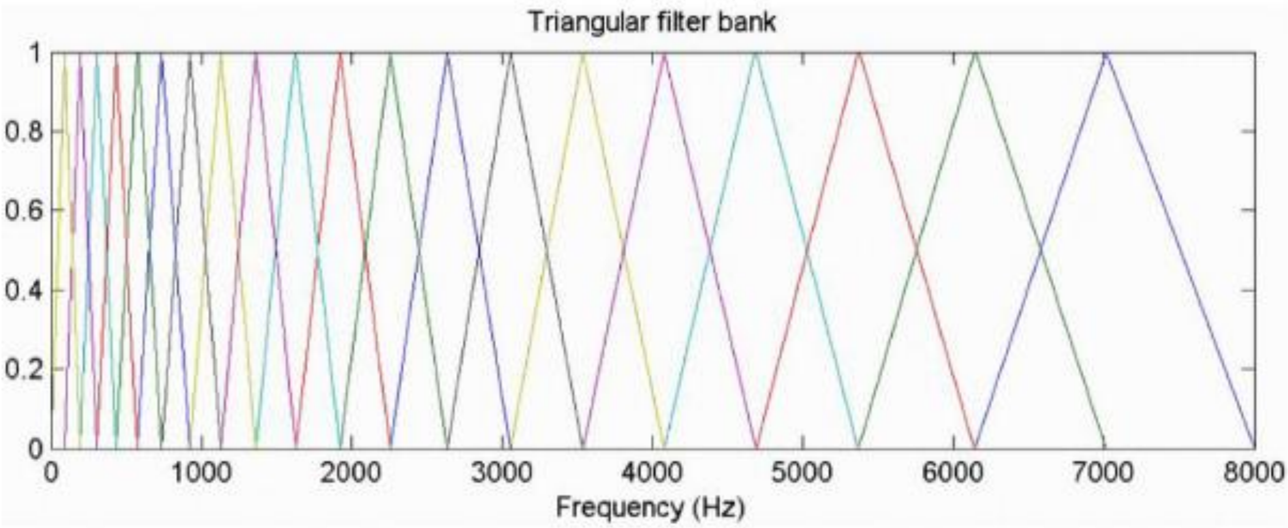
MelFilterbank
Mel filterbank computes a set of ***MFCC*** for a generated set of filters and spectral density coefficients of given frame.
1. Generation of filters:

- For a given frequency range in Mel scale and n amount of filters, *n + 2* linearly spaced filter intervals are generated
- Filter intervals are converted back to Hertz scale.
- Corresponding to the number of coefficients of ***FFT*** used in **Periodogram,** *n + 2* filter intervals are generated.

  $$F_i = \lfloor \frac{(2 \cdot L + 1) \cdot f_i}{S_r} \rfloor$$

  L - number of coefficients of ***FFT*** f_i - *i*-th filter interval in Hertz scale
  S_r - sample rate of the recording
- Finally, a filterbank of *n* vectors of triangular filters is computed according to the below formula:
  Formula for obtaining *k*-th filter*:*
  For *i* from *0* to *max(F)* where *F* is set of ordered filter intervals:

$$Filter_k(i) = \begin{cases} 0 & i < F_{k-1} \lor i > F_{k+1} \\ \frac{i - F_{k-1}}{F_k - F_{k-1}} & F_{k-1} \leq i \leq F_k \\ \frac{F_{k+1} - i}{F_{k+1} - F_k} & F_k \leq i \leq F_{k+1} \end{cases}$$



2. Generation of ***MFCC***
- For each filter in a filterbank, the energy in the filter is obtained by the dot product between a filter vector and a vector of spectral estimates of a single frame.
- Because humans perceive loudness in logarithmic scale, logarithm of filter energies is taken.

  $E_k = \ln(Filter_k \circ P_{frame})$ P_{frame} - spectral density estimates vector

  E_k - energy of *k*-th filter

Finally, a ***DCT*** of energies of a single frame is computed, resulting in a set of *n* real valued ***Mel Frequency Cepstral Coefficients***.

SpeechFeatureExtractor

Speech Feature Extractor follows the steps below to obtain a feature vector for a single voice recording.

1. Framing of the signal:
- Speech Feature Extractor feeds a vector of signal samples into the **FrameMaker**, obtaining back a set of equally spaced intersecting frames.

2. Windowing the signal:
- In preparation for ***FFT**, to each frame a window (**Gauss** or **Hamming**) function is applied.*

3. Obtaining spectral estimates by **Periodogram**:
- Each frame is subjected to ***FFT*** and a set of spectral density estimates is obtained.

4. Creation of a **Mel filterbank**:
- For given length of ***FFT*** and sample rate, a filterbank of triangular filters is created.
- For each frame, its set of spectral density estimates is filtered through the filterbank, obtaining a set of energies left in each filterbank.
- A ***DCT*** of logarithms of energies is computed, resulting in a set of ***Mel Frequency Cepstral Coefficients***.

5. Combining the ***MFCC*** sets:

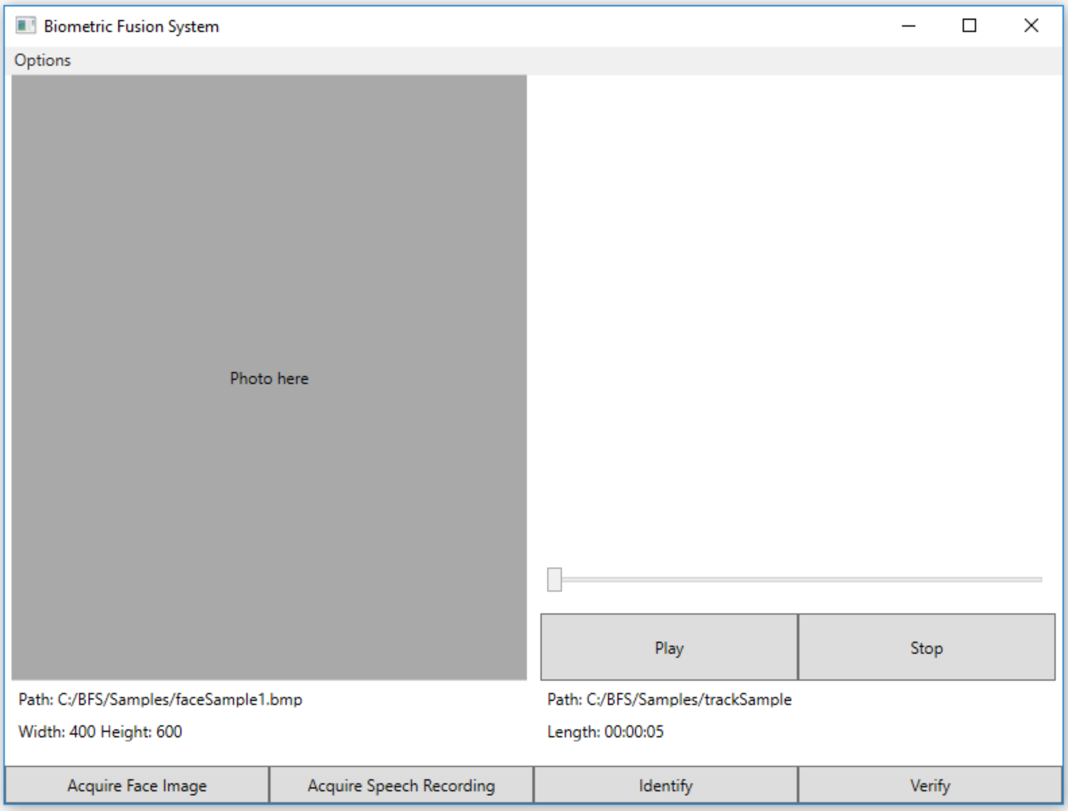
Sets of ***MFCC*** of each frame are concatenated, resulting in a feature vector of the recording.

DynamicTimeWarping
Dynamic Time Warping is used for comparing two sequences of elements that differ in length. It returns the sum of distances between elements of those two sequences.
Pseudocode:

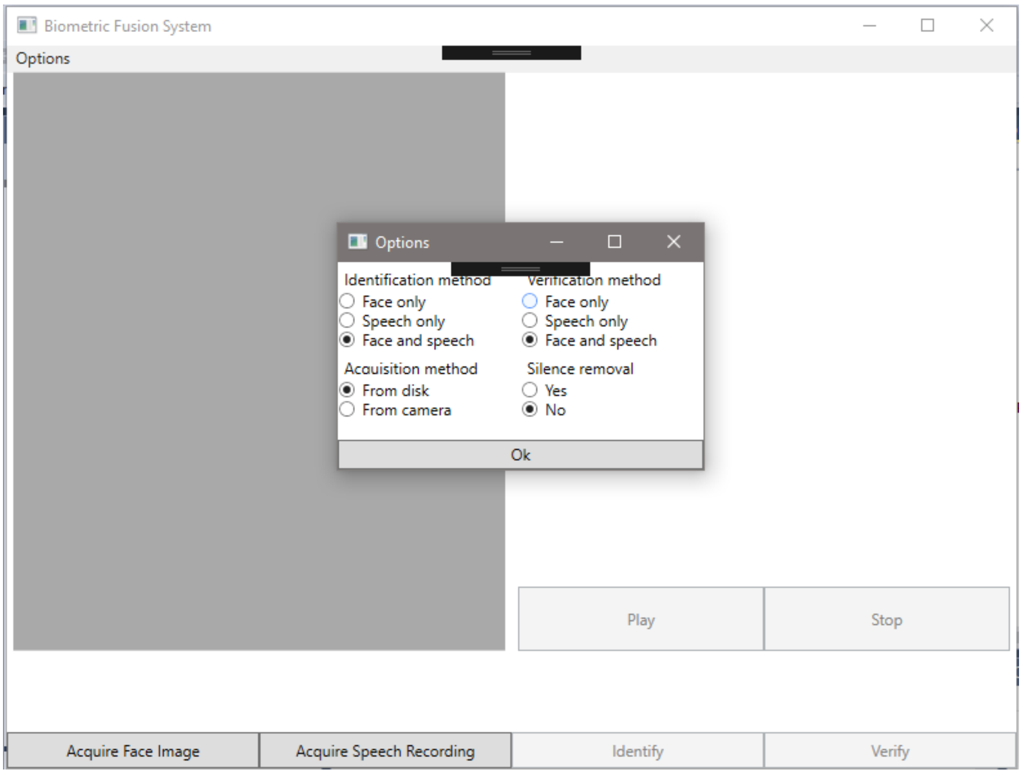
```
int DTWDistance(s: array [1..n], t: array [1..m]) {
  DTW := array [0..n, 0..m]
  for i := 1 to n
      DTW[i, 0] := infinity
  for i := 1 to m
      DTW[0, i] := infinity
  DTW[0, 0] := 0
  for i := 1 to n
      for j := 1 to m
          cost := d(s[i], t[j])
          DTW[i, j] := cost + minimum(DTW[i-1, j  ],    // insertion
                                      DTW[i  , j-1],    // deletion
                                      DTW[i-1, j-1])    // match

  return DTW[n, m]
}
```

3.5 Graphical User Interface

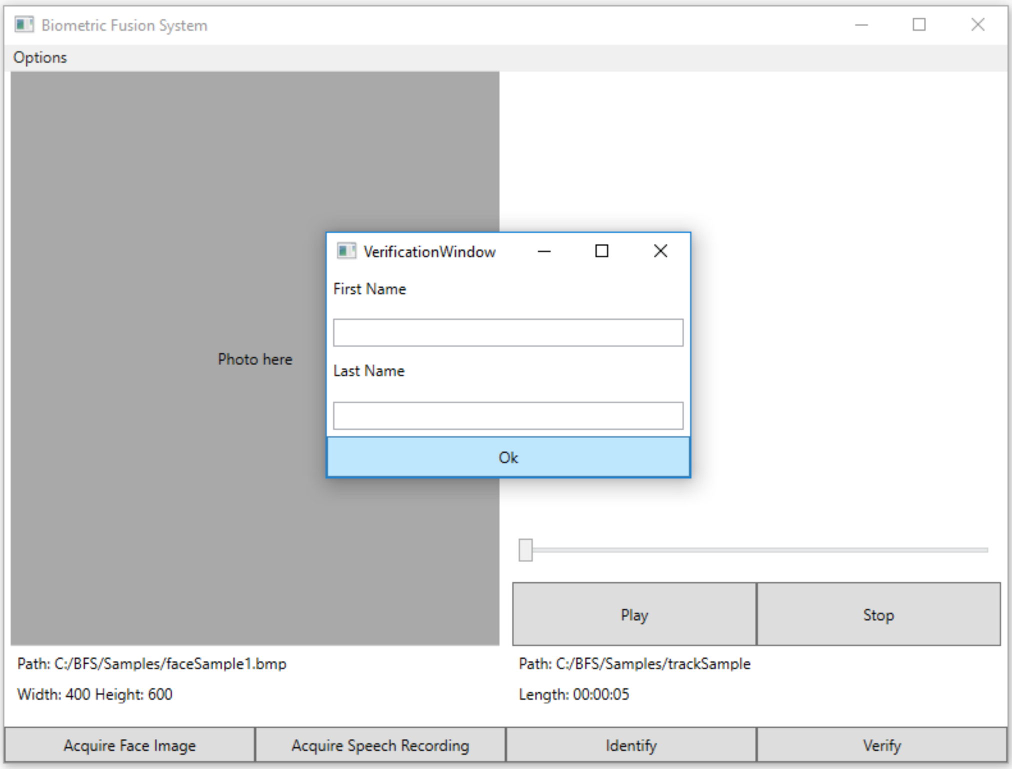
The main window of an application:



The options window:



The verification window:



4 Tests results

The project is to explore the results of different human recognition algorithms to maximize the success rate of identification of VP.

4.1 Early tests results

The dataset used for these tests consists of two persons – 8 speech recordings and 8 face images (4 for each person). Each of the aforementioned are used for the respective tests.

4.1.1 Early speech tests results

Current implementation for the speech recognition algorithm uses the MFCC and DTW. When applied to the input recordings which are compared to the templates stored in the database program achieved a percentage success rate of 70%. After removal of the silence from the recordings, the program increased the success rate to 83%.

4.1.2 Early face tests results

Current implementation for the face recognition algorithm uses the Gabor filter and color histogram in order to derive the feature vectors and the MDC classification method. When applied to the input images which were compared to the templates stored in the database program achieved a percentage success rate of 100%.

4.2 Datasets description

The samples in the dataset consist of recordings of two words: 'algorithm' and 'close', and photos of faces. Training samples are composed of four recordings of each word and four photos of face of a given person. Test samples are composed of two recordings of each word and two photos of face of a given person.

- Small dataset description:

Small dataset consists of 21 people, where each person provided both training and test data.

- Medium dataset description:

Medium dataset consists of 40 people, where 21 provided both training and test data, and 19 provided only training data.

- Large dataset description:

Large dataset consists of 58 people, where 39 provided both training and test data, and 19 provided only training data.

Every person present in the small dataset is also present in the medium and large datasets.

4.3 Final identification tests results

Below presented are the results for identification of a person among the people stored in the database, and the parameters used for face and voice identification algorithms.

4.3.1 Voice recognition parameters description

We have chosen 5 parameters to be adjusted for voice recognition:

-length of a single frame of the recording
-interval between each frame of the recording
-number of filterbanks used in calculation of MFCCs
-number of Mel Frequency Cepstral Coefficients taken from extraction
-type of a window function applied to the recording samples

By the method of trial and error, we have chosen following values of parameters for best quality of voice feature extraction:

-length of a single frame of the recording: 0.03 seconds.
-interval between each frame of the recording: 0.015 seconds
-number of filterbanks used in calculation of MFCCs: 18 filterbanks
-number of Mel Frequency Cepstral Coefficients taken from extraction: 11 coefficients
-type of a window function applied to the recording samples: Hamming window function

Result of small dataset speech identification:
For recorded word 'algorithm', 73.81 % (31 test samples out of 42, 21 persons identified, 21 persons in database) were correctly identified.
For recorded word 'close', 90.48 % (38 test samples out of 42, 21 persons identified, 21 persons in database) were correctly identified.

Result of medium dataset speech identification:
For recorded word 'algorithm', 71.43 % (30 test samples out of 42, 21 persons identified, 40 persons in database) were correctly identified.
For recorded word 'close', 88.10 % (37 test samples out of 42, 21 persons identified, 40 persons in database) were correctly identified.

Result of large dataset speech identification:
For recorded word 'algorithm', 71.79 % (56 test samples out of 78, 39 persons identified, 58 persons in database) were correctly identified.
For recorded word 'close', 70.51 % (55 test samples out of 78, 39 persons identified, 58 persons in database) were correctly identified.

4.3.2 Face recognition parameters description

We have chosen 4 parameters to be adjusted for face recognition:

-input wave length of Gabor filter
-number of orientations (angles) of Gabor filter
-horizontal standard deviation of Gabor filter
-vertical standard deviation of Gabor filter

By the method of trial and error, we have chosen following values of parameters for best quality of face feature extraction:

-input wave length of Gabor filter: 4 pixels
-number of orientations (angles) of Gabor filter: 7
-horizontal standard deviation of Gabor filter: 6
-vertical standard deviation of Gabor filter: 7.5

Result of small dataset speech identification:
Out of all the faces, 64.29 % (27 test samples out of 42, 21 persons identified, 21 persons in database) were correctly identified.

Result of medium dataset speech identification:
Out of all the faces, 57.14 % (24 test samples out of 42, 21 persons identified, 40 persons in database) were correctly identified.

Result of large dataset speech identification:
Out of all the faces, 47.44 % (37 test samples out of 78, 39 persons identified, 58 persons in database) were correctly identified.

4.3.3 Speech tests results

SPEECH IDENTIFICATION RESULTS					
DATASET	**WORD**	**SUCCESS RATE IN %**	**SUCCESS RATE OF TEST SAMPLES**	**AMOUNT OF IDENTIFIED PERSONS**	**DATASET SIZE**
SMALL	**ALGORITHM**	**73.81%**	**31/42**	**21**	**21**
SMALL	**CLOSE**	**90.48%**	**38/42**	**21**	**21**
MEDIUM	**ALGORITHM**	**71.43%**	**30/42**	**21**	**40**
MEDIUM	**CLOSE**	**88.10%**	**37/42**	**21**	**40**
LARGE	**ALGORITHM**	**71.79%**	**56/78**	**39**	**58**
LARGE	**CLOSE**	**70.51%**	**55/78**	**39**	**58**

4.3.4 Face tests results

FACE IDENTIFICATION RESULTS				
DATASET	SUCCESS RATE IN %	SUCCESS RATE OF TEST SAMPLES	AMOUNT OF IDENTIFIED PERSONS	DATASET SIZE
SMALL	64.29%	27/42	21	21
MEDIUM	57.14%	24/42	21	40
LARGE	47.44%	37/78	39	58

4.4 Final verification tests results

Below presented are the results for verification of a person against the given person stored in the database.

4.4.1 Speech tests results

SPEECH VERIFICATION RESULTS					
DATASET	WORD	FALSE ACCEPTANCE RATE IN %	FALSE REJECTION RATE IN %	AMOUNT OF VERIFIED PERSONS	DATASET SIZE
SMALL	ALGORITHM	14.40%	23.81%	21	21
SMALL	CLOSE	19.29%	19.05%	21	21
MEDIUM	ALGORITHM	13.49%	23.81%	21	40
MEDIUM	CLOSE	22.95%	19.05%	21	40
LARGE	ALGORITHM	11.74%	19.23%	39	58
LARGE	CLOSE	30.48%	15.38%	39	58

4.4.2 Face tests results

FACE VERIFICATION RESULTS				
DATASET	FALSE ACCEPTANCE RATE IN %	FALSE REJECTION RATE IN %	AMOUNT OF VERIFIED PERSONS	DATASET SIZE
SMALL	31.31%	4.76%	21	21
MEDIUM	26.37%	4.76%	21	40
LARGE	25.17%	3.85%	39	58

5 User manual

This chapter is to help the user understand and use the system.

5.1 User guide

Start-up guide:

In order to launch the application the user has to have the following on the workstation:

- Application.exe
- Microsoft SQL Server version 2012 or newer installed

User guide:

For the user to be able to identify or verify a person, a picture of face and/or a recording of speech have to be loaded into the program. To specify which of the aforementioned is used, user has to select the respective options in options window – "Face only", "Speech only" or "Face and Speech".

In order to load the picture to the application, the user has to press the "Acquire Face Image" button and select a picture from disc or use the camera to capture it. The choice between the methods of acquiring the image is set in the options window in the left upper corner of the GUI. Picture is shown on the left hand side of the application.

For the acquisition of the speech, the user has to press the "Acquire Speech Recording" button. Analogically to the face image, the user can choose between two methods of obtaining the recording – load from the computer's hard drive or record it using the microphone built in the camera. Voice recordings graph is visible on the right hand side of the application. Furthermore, the user can play the recording in the application.

Additional option in the options window is the "Silence removal". If selected, the application removes the silent parts of the recording loaded into the application. This is done upon loading the recording.

Both face picture and speech recording have their absolute file path listed below, as well as the images height and width and the length of the recording.

If the data is loaded correctly, the user has to select the method he wishes to use – identification or verification, by pressing the respective "Identify" or "Verify" button.

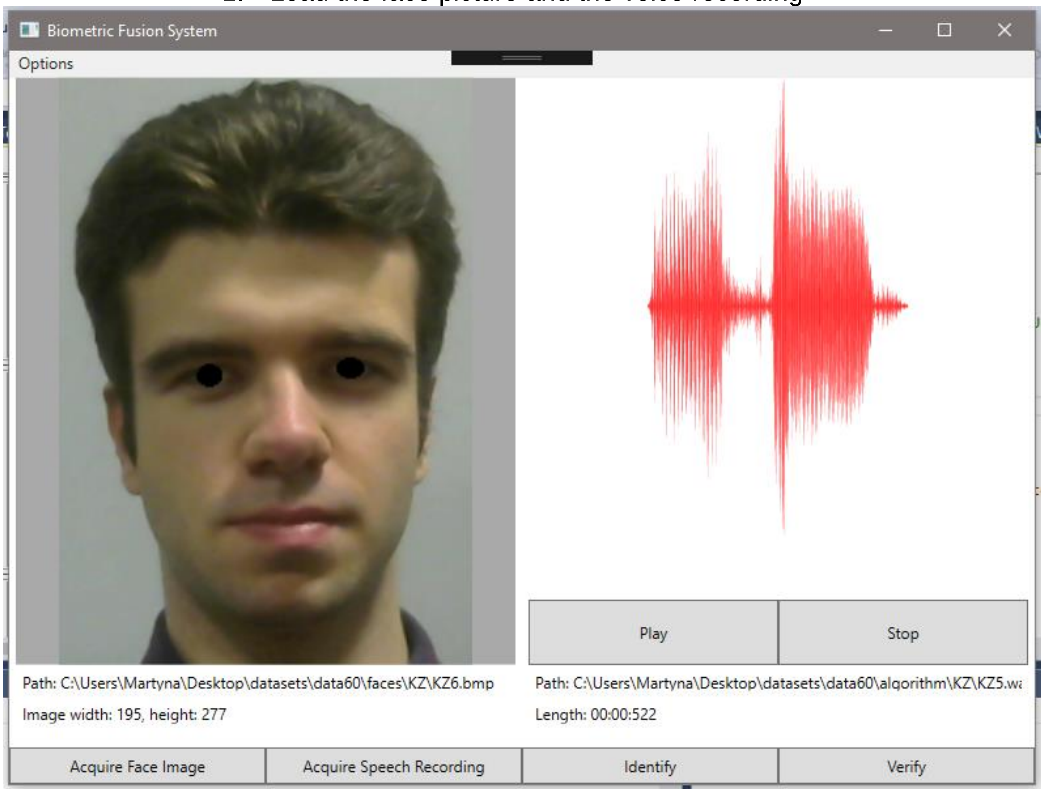
Selecting the identification matches the input data against the templates stored in the database. After this process is finished, a message window appears and inform the user what is the result. If both speech and face were selected, the application returns the results for both methods separately by writing the names of the people recognized in the input data.

Selecting the verification triggers a pop-up window where user has to fill in the first name and the last name of the person (must be previously loaded in the database) for whom he claims to be. Then, the input data is compared with the template stored in the database. When finished, a message window appears informing whether the data was recognized or not. If both speech and face were selected, the application returns the results for both methods separately by writing true or false if a person was recognized or not.

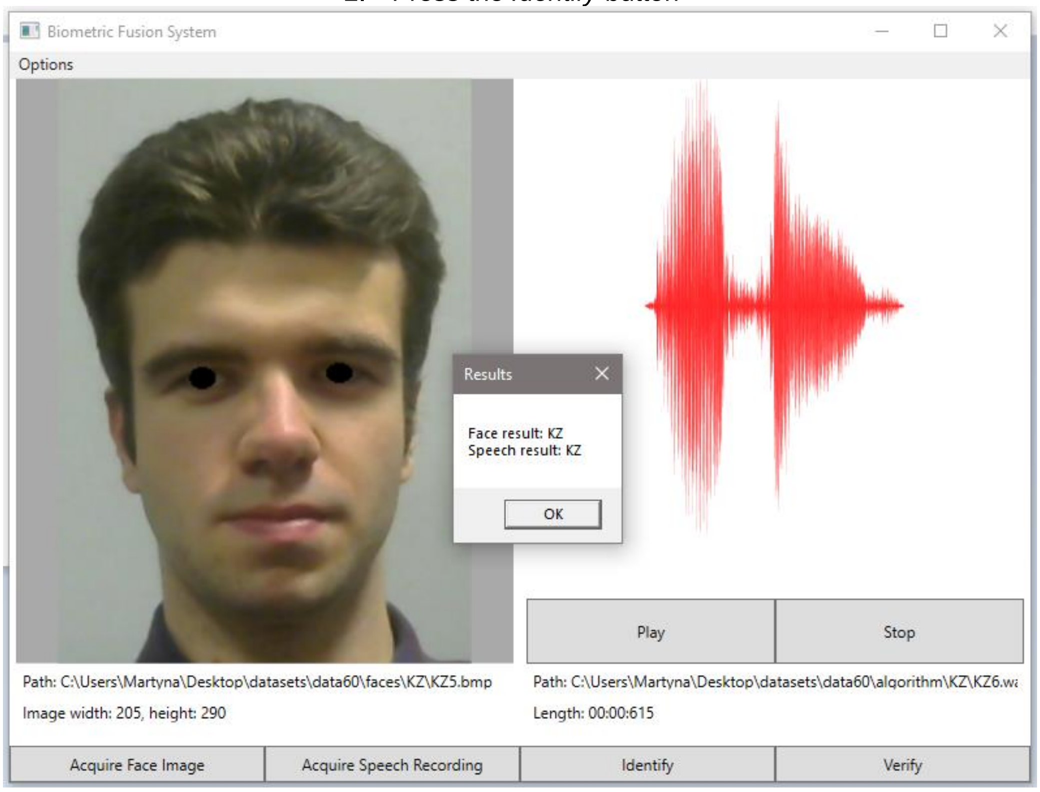
5.2 User stories

- Identification of a person

    1. Load the face picture and the voice recording

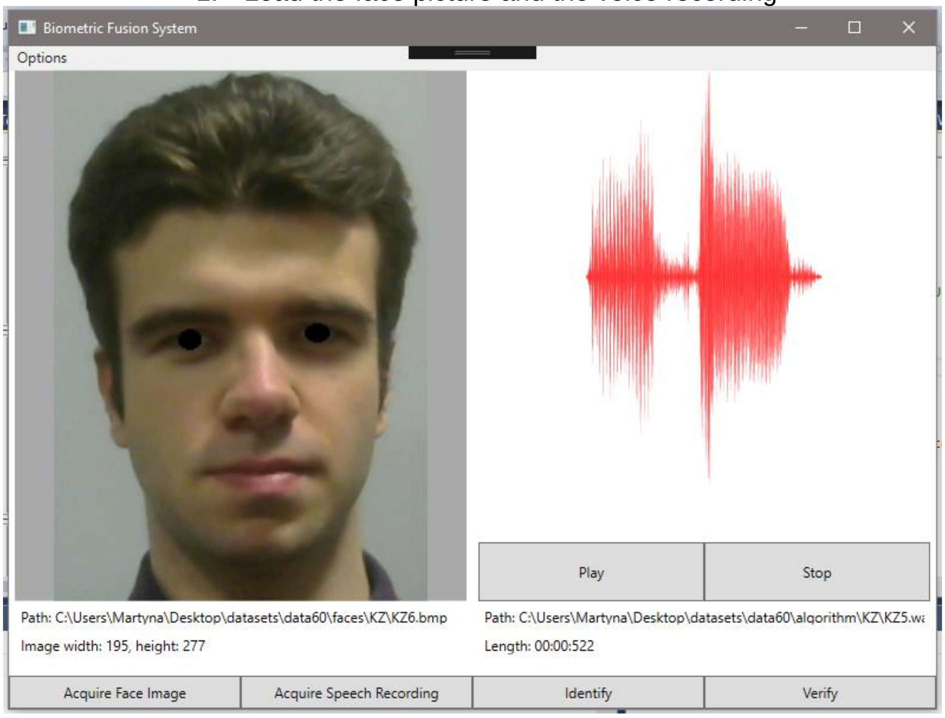


    2. Press the *Identify* button

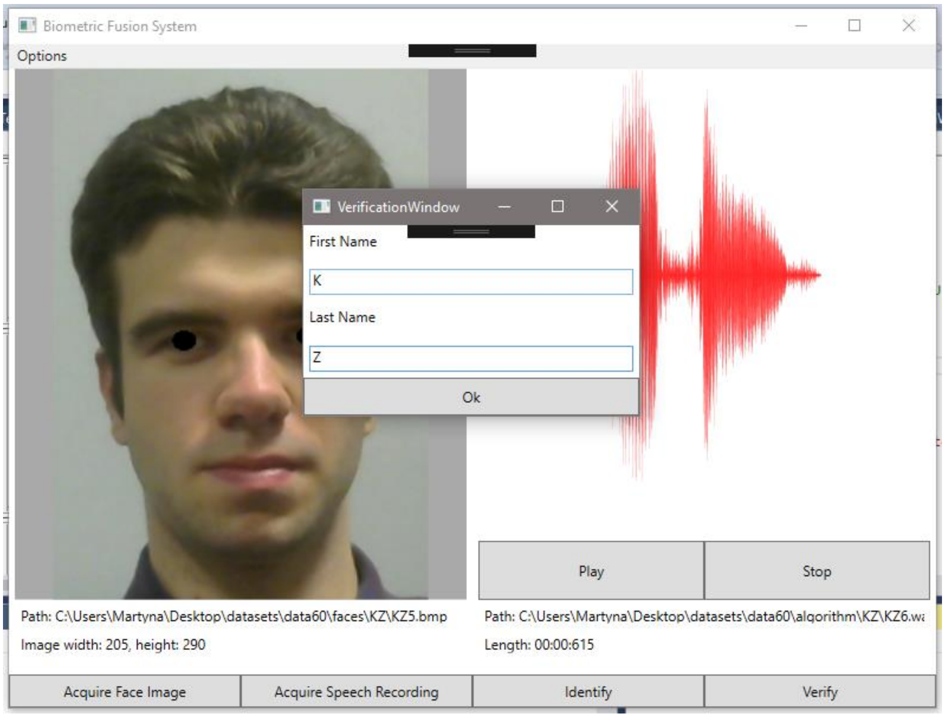


- Verification of a person

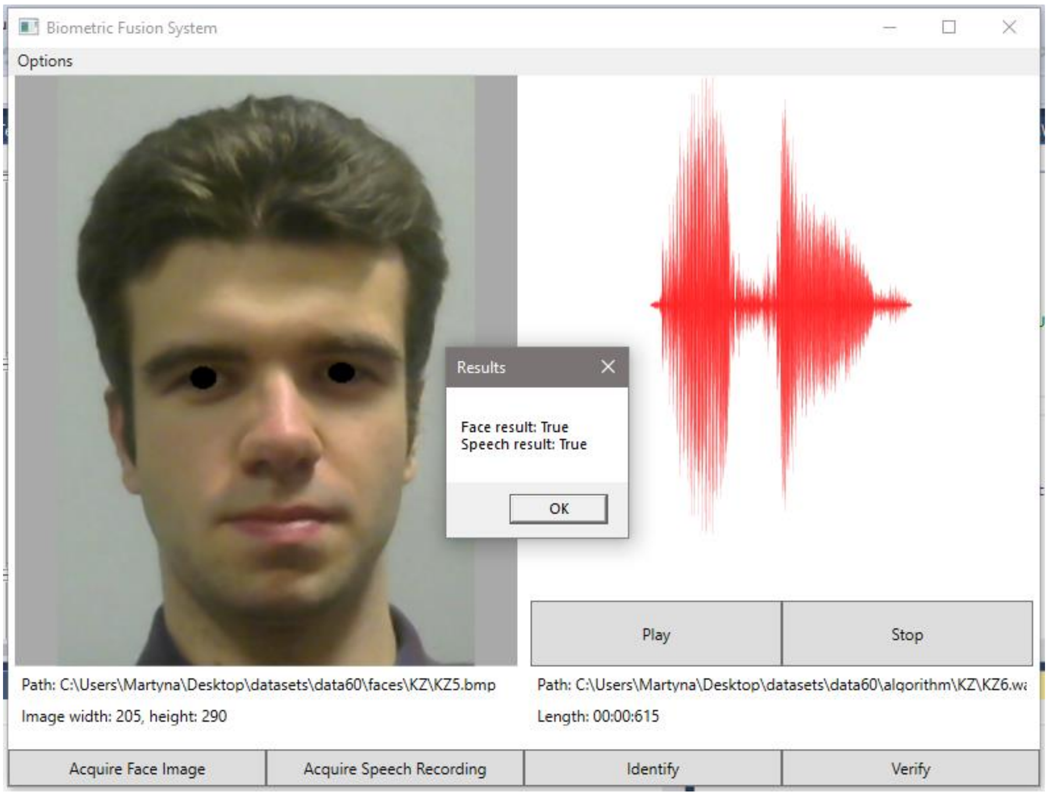
1. Load the face picture and the voice recording



2. Press the *Verify* button and input the first name and last name of a person to whom the loaded data belongs



3. Press *OK*


Biometric Fusion System
Options
Results
Face result: True
Speech result: True
OK
Play
Stop
Path: C:\Users\Martyna\Desktop\datasets\data60\faces\KZ\KZ5.bmp
Image width: 205, height: 290
Path: C:\Users\Martyna\Desktop\datasets\data60\algorithm\KZ\KZ6.wa
Length: 00:00:615
Acquire Face Image
Acquire Speech Recording
Identify
Verify

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Minimum Distance Classifiers tutorial link:
https://homepages.inf.ed.ac.uk/rbf/HIPR2/classify.htm (accessed at 06.12.2017)

FFmpeg official website link:
https://www.ffmpeg.org/ (accessed at 06.12.2017)