UNIVERSITY OF EAST ANGLIA

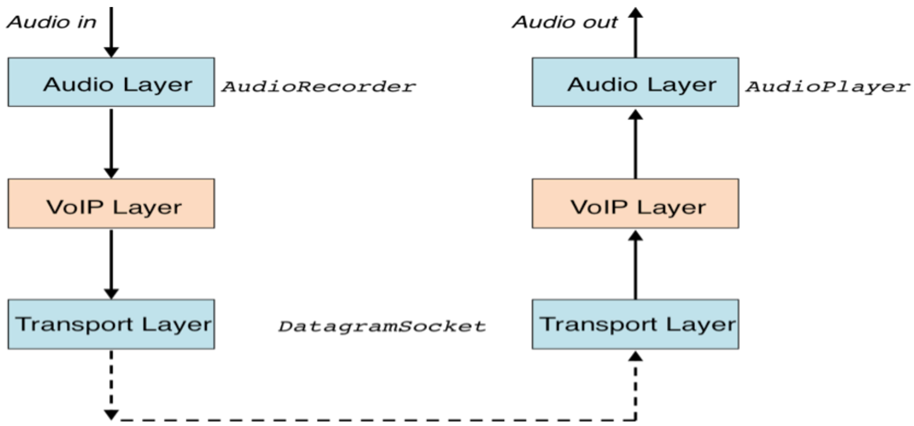
School of Computing Sciences

CMP-6009B Coursework 1

ASSIGNMENT TITLE:	**Design, implementation and evaluation of a VoIP communication system**
DATE SET:	**Week 3**
DATE & TIME OF SUBMISSION:	**Report – Monday Week 8, Demonstration – Thursday/Friday Week 7**
RETURN DATE:	**Week 10**
ASSIGNMENT VALUE:	**30%**
SET BY:	**BPM SIGNED:**
CHECKED BY:	**GI SIGNED:**

Description of assignment:

The aim of this assignment is to design, implement and evaluate a VoIP communication system to operate between two PCs in the Lewin Lab. The assignment is to be tackled in pairs – typically with one person responsible for the sender and the other for the receiver. This brings in issues such as the standardisation of VoIP packets. The design should follow a layered approach with a suitable structure shown below.



The audio layer is provided by the `AudioRecorder` and `AudioPlayer` classes that are introduced in Lab 1. The transport layer should use the `DatagramSocket` class in Java as the UDP access point which is covered in Lab 2. **The task of this coursework is to design and implement the VoIP layer.**

A VoIP layer will need to be developed for the sending machine and the receiving machine. In its most basic operation the VoIP layer on the sender will need to take audio blocks from the audio layer and pass them down into the transport layer. On the receiving side the VoIP layer will receive packets from the transport layer and will need to pass them to the audio layer ready for playback.

Such a system should work well under ideal network conditions. However, in this coursework three other `DatagramSocket` classes will be provided which simulate three non-ideal channel conditions:

- `DatagramSocket2`
- `DatagramSocket3`
- `DatagramSocket4`

In order for your VoIP system to operate effectively under these three unknown channels you must first analyse the packet loss characteristics of the three channels and then design optimal systems for each channel. The analysis could be achieved by sending packets across the network and monitoring what is received.

Once VoIP systems have been developed for the four network conditions (`DatagramSocket`, `DatagramSocket2`, `DatagramSocket3` and `DatagramSocket4`) an evaluation of the resulting Quality of Service should be made. This should include parameters such as bit rate, delay, packet efficiency and a measurement of speech quality. You should attempt to maximise QoS for each channel.

Assessment criteria:

The assessment is in two parts:

1. Practical demonstration of VoIP system

This will be made to BPM and takes the form of a demonstration of the VoIP system under the four different network conditions (`DatagramSocket`, `DatagramSocket2`, `DatagramSocket3` and `DatagramSocket4`). Marks will be awarded for functionality such as full-duplex connection, the effectiveness of error concealment, delay minimisation and quality of the speech. Both team members will participate in the presentation.

2. Technical report

This should describe results of the network analysis carried out to determine the conditions of the channels (`DatagramSocket`, `DatagramSocket2`, `DatagramSocket3` and `DatagramSocket4`).). Design decisions of the VoIP system should be explained and implementation discussed. Results of QoS testing should be presented and include parameters such as bit rate, delay, packet efficiency and speech quality.

Required:

1. Send details of pairings to b.milner@uea.ac.uk by the end of **week 3**.
2. Short (informal) discussion on the results of network analysis into the channel conditions (`DatagramSocket`, `DatagramSocket2`, `DatagramSocket3` and `DatagramSocket4`) and initial design ideas. This will take place during the lab in week 4 or week 5 and is a **formative assessment** to give feedback on your initial ideas.
3. Practical demonstration of VoIP to take place in the Lewin Lab on Thursday/Friday of week 7. A sign up sheet will be provided for each group to book a slot in advance. Each group will have 15 minutes to demonstrate their VoIP system on `DatagramSocket`, `DatagramSocket2`, `DatagramSocket3` and `DatagramSocket4`.
4. Technical report. This is a joint report that should be no longer than 5 sides of A4. The report should include the following three areas: i) discussion of the network analysis carried out, ii) description and justification of the design of the VoIP system, iii) an evaluation of the final system in terms of its QoS.

Allocation of marks:

Demonstration (50%)

Quality of service under no packet loss – `DatagramSocket` (15%)

Quality of service with `DatagramSocket2` (15%)

Quality of service with `DatagramSocket3` (10%)

Quality of service with `DatagramSocket4` (10%)

Report (50%)

Network analysis (10%)

Design and implementation of the VoIP system (20%)

Evaluation of VoIP system in terms of QoS parameters (15%)

Structure and technical writing style (5%)

Learning outcomes:

- Be able to transmit IP packets between two host computers using the `DatagramSocket` class in Java
- Be able to record and play back audio data on a PC using Java
- Be able to put these two components together to implement a simple VoIP communication system
- Be able to measure the characteristics of a network and the resulting QoS of the VoIP system
- Know the problems associated with unreliable network communication when delivering real-time services and be able to design and implement methods to compensate for these problems
- Know that standards are crucial for practical deployment of VoIP
- Work as a small team to agree on design decisions and implement a working system