CMP-5037B Networks

*Design, implementation and evaluation of a secure VoIP communication system*

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Network Analysis

DatagramSocket

DatagramSocket2

DatagramSocket3

When sending packets across the DatagramSocket3 channel, it was clear that the issue of packet loss was being simulated here, as there was a noticeable degradation of audio quality in the data being received. Therefore, we came to the conclusion that our VoIP program would have to implement a compensation system for lost packets. To research this further, we put together an advantages and disadvantages table for sender-based compensation and receiver-based compensation, to decide which one would be better to use.

**Sender-based compensation**

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| **Advantages** | **Disadvantages** |
| **Better audio quality** – packets are re-sent when lost | **Slow** – for a real-time system, waiting for a packet to be re-sent is **highly impractical** |
| **Reliable** – acknowledgements are set to determine whether a packet reached the receiver or not |  |

**Receiver-based compensation**

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| **Advantages** | **Disadvantages** |
| **Faster –** concealment of occurrences of packet loss can be applied much more quickly | **Audio quality is affected –** a degradation in audio quality is noticeable when using techniques such as splicing |
| **Good for short bursts of audio** – techniques such as repetition or using a fill-in packet are effective for short bursts of audio |  |

Through this analysis, it became clear that using receiver-based compensation, specifically repetition, would be the most appropriate for our system. It’s much faster than server-based compensation, which is crucial as speed is essential in a real-time system. Furthermore, it’s effective when only a small number of packets are lost in succession, making it easy to conceal this effect for short bursts of audio.

DatagramSocket4

When sending packets across the DatagramSocket4 channel, we noticed a clear decrease in good QoS, jitter causing an intermittent static noise to be heard as packets were received. Realising that our system would have to try and compensate for this, we did some research into jitter buffering, a technique used in some VoIP systems to minimise the effect of jitter on the QoS.

A jitter buffer works by intentionally delaying incoming packets, so they can be temporarily stored and re-ordered before the audio data is played. With the packets then being in the correct order, better audio quality is preserved.

VoIP System Design

Security

QoS evaluation

Project management