ECE3141 Information and networks short projects

Introduction

Students working in pairs will undertake a short project. The projects will involve writing MATLAB or other simulation programmes or perhaps analysing the outputs of network analysis tools, and communicating the results of the investigation in a short report and a short (5 minute) video presentation on the topic. The current online learning environment we all find ourselves in means that the way you will present your work will of necessity be different to how we would ordinarily do this. We will provide precise details and confirm this closer to the submission deadline, but you can expect the following:

- Completion and submission by the end of week 10
- Submission in the form of a 2-page written report (details below) plus a 5-minute recorded video presentation. (Not live – you record this when you want and upload it by the submission date – see later in this document for more details.) Each student is expected to speak for about equal time in this video presentation.
- You will choose a time slot during the time period of your regular lab session during the last two weeks of the semester to join a short (just a few minutes) Zoom between the lecturer, you and your partner to get you to answer a couple of questions and to allow you to demonstrate that BOTH students in the pair have a solid understanding of the topic, analysis and conclusions.
- While students are expected to define the project objectives and experiments themselves (see below), we recognise that in these days there is little opportunity for one-on-one discussion about these topics so there will be some drop-in zoom meeting sessions organised for you to clarify something if you are lost and confused.

The projects are worth 13% of the total assessment for the unit. This is equal to all the lab classes combined, so you are expected to put in a commensurate amount of effort.

Aims of the projects

By undertaking these projects, students will

- Develop an understanding of a specialised aspect of telecommunications,
- Gain experience in using MATLAB and the functions available within the Communications Toolbox, or other analysis tools and techniques
- Develop skills in exploring a technical topic without clearly defined objectives.
- Develop skills in writing technical reports,
- Develop skills in presenting talks on technical topics
- Develop skills in collaboration with colleagues to achieve an agreed objective.

Note that, while some of the projects use tools and concepts discussed in the lectures, others do not. The aim of the project is not to reinforce lecture material, but rather to allow you to explore a telecommunications topic, report on and present it.

Even if we feel we are living in unusual times with everything being online, it could be argued that global distancing and the remote working and learning we are experiencing is just going to accelerate a move towards online collaboration generally. That is, after the pandemic is controlled, we may still see a sustained shift to online work and online collaboration with colleagues and team members. You may have to overcome some challenges working collaboratively with your partner when you can only contact them electronically, but these are likely to be very important skills in the years ahead.

Organisation

Students will work in pairs. Students can choose their own partners from within the same lab session. Student pairs will be able to select their topic from a list (see the pre-defined projects at the end of this document), or they can propose a topic (within constraints – see below). When you select a topic, you will also nominate a time slot in the week 11 or 12 lab class time in which to answer some questions via zoom (see below). There is a limit on the number of student pairs in a lab class who can do the same topic, so those who organise themselves fastest will have most choice. Note that any topic proposed by students must be approved by the lecturer in charge. Keep in mind that the project must involve more than just a literature search; there must be experimentation, data collection, analysis and interpretation.

Details about the registration procedure will be advertised on Moodle very soon.

Report

Each pair must write a short report on their topic. This should be written in your own words. Any sources (including any copied graphics used for figures) must be properly referenced. Use the IEEE style of referencing – see http://www.lib.monash.edu.au/tutorials/citing/ieee.html. The reports are due with the recorded video presentations at the end of week 10 (Friday 29th May). Specific details will be provided on Moodle before then.

The main body of the report should be a maximum of two A4 pages (that is, two SIDES, not two sheets) using 12 point font. The main body must include at least one block diagram describing your topic, at least one graph or diagram illustrating your results and at least one reference to a relevant book or paper. (You can also include references to Web sites but at least one reference must be to a published book or paper.) Programme listings should be provided as appendices. If any programme statement is copied from any source (including the Matlab Help files) these must be clearly labelled with comments which acknowledge the source. If you believe that your particular project demands additional diagrams, plots or results that will not fit in the main body, then these, too, may be placed in an appendix. (The main body still has to include at least one graph or table of results, though.) Add such extra material ONLY if it is essential to explain your key results; do not just include lots of plots to try to make it look like you were busy. It will be a struggle to fit what you'd like to say in just two pages, but this is part of the exercise; it is important to decide the right information to include.

Video presentation

You can use whatever video recording and editing programmes you like, but we need to see each speaker at least at the start introducing themselves. The preferred format is a recording of the two group members in a Zoom meeting. The minimum you should aim to achieve is the following:

1. First student, full screen, introducing themselves
2. Second student, full screen, introducing themselves
3. Continue rest of presentation using powerpoint (or similar) presentation and alternating speakers as you prefer.

If you are able to record you and your partner in a Zoom meeting, with the thumbnail videos maintained at the side as you present other graphical material by screen-share, this would be ideal and involve little or no extra work in editing. Our understanding is that you should have the necessary permissions to allow recording using Zoom.

If Zoom recording is not possible, you could consider other options such as:

- For the Mac, it's relatively straightforward. You can use Quicktime to do a Screen recording or Movie recording, and it can also be used to join clips together. Or, if you want, you can do more sophisticated editing with iMovie.
- For Windows, there are numerous free or free-trial screen recording and video editing solutions[1].

While we want you to take some care in presenting your work, please do not spend all the time on video editing and highly refined scene transitions; you will certainly earn more marks if you devote that time to the theory, analysis and interpretation of your project topic. Do not under any circumstances submit two separate videos from the two students; it would suggest you've completely failed to collaborate on the project, and this is one of the main requirements.

It is preferable that you draw your own figures or animations that you use in your talk using drawing tools such as those in Powerpoint. The use of figures or animations from other sources is discouraged but, if you choose to use these, the source(s) must be clearly acknowledged.

The video presentation will be five minutes long, with an allowable variation of 30 seconds (i.e. 4:30 – 5:30 is the acceptable range; you will lose marks outside this range, or if you clearly don't have material to fill this time but try to "pad it out" instead). Both students in each pair must present some of the talk for approximately equal times. It's up to you if you want to devote the first half to one student and the second half to the other, or if you want to "tag-team" (switch repeatedly between speakers throughout the presentation). You MUST introduce yourselves individually at the start of the presentation, so we are clear about who is talking.

Follow-up Q&A

During weeks 11 and 12 (in Malaysia, week 11 only), in the time slot you nominate when you choose your topic, there will be a brief (couple of minutes) zoom meeting between both students and the lecturer, during which you'll be asked to answer a couple of questions or discuss some aspect of your work. This is intended to probe whether you really do understand what you've done and to make sure that BOTH students have contributed.

Assessment of short projects and expected level of achievements

The projects will be assessed based on a combination of the technical content, demonstrated understanding and ability to communicate the work presented as demonstrated in the report, presentation video and responses to questions.

Normally both students in a pair will receive the same mark, unless the standard of presentation of one student is very different from the other, or the response to questions indicates a substantial inequality of understanding or contribution to the project. It is important that you choose a partner who you believe will contribute equally to the task. You are generally expected to sort out any difficulties of working together, and only in the most extreme case bring a problem to the attention of the lecturer.

It is impossible to separate the marks for the technical content and for the presentation of

[1] See, for instance, https://www.howtogeek.com/355524/how-to-use-windows-10s-hidden-video-editor/

the project results. For example, if the report and presentation are presented poorly, it may be difficult to determine the quality of the technical work. Similarly, it is much more difficult to write a clear report and give a talk on work with substantial technical content, than it is to give a talk on superficial matters. As a result, you will get a single mark that represents the overall quality of your work as presented.

Each student will receive a mark and a short, written comment on their project (via Moodle). Due to the number of reports and the workload involved, it will not be possible to provide detailed feedback.

Each of the projects is open-ended. There is no single correct answer. Examples of the type of performance which will earn different marks are given below.

Level of technical contribution and presentation	**Mark/10**
Students demonstrate clear understanding of their topic and demonstrate significant initiative in their solution. The programmes are clearly written and documented, with sensible, understandable variable names. The report meets all of the criteria for length, etc. and is clearly written and grammatically correct. The presentation is very clear, the students talk clearly, the slides are well prepared and make good use of diagrams, etc to illustrate the key points. The talk ends on time. The students can answer questions clearly and convincingly.	10
Students demonstrate a clear understanding of their topic and the work presented meets that specified in the short project description. The report meets all the criteria for length, etc., but may have a few minor errors. The talk is clear and the slides are well prepared and make use of diagrams to illustrate the points. Students can answer the questions asked, but do not demonstrate a thorough understanding of the topic.	7.5
Students demonstrate an understanding of the main points of the topic. The work almost meets the specification. The report or video have some limitations. For example, it may be over-length or have poor or confusing explanations. Students can answer simple questions but cannot answer more difficult ones.	5
Students do not understand the topic and present trivial /irrelevant results which do not meet the specification. The talk is clear and the presentation meets the guidelines for length, etc., but does not have adequate technical content.	3

Note that the mark (out of 10) will determine 13% of your total assessment for this subject; the same as for all the laboratories combined. You should take this into account when you decide how much effort and time to devote to it. It is our experience that it is very obvious when students decide to leave the project until the last minute and then rush through in a couple of hours trying to come up with something that looks convincing; it is not convincing.

Hint: A major challenge in presenting what could be a lot of information in very few written words, or very few presentation minutes, is to choose the right information. If we are very proud of the code we have written or the problems we overcame in getting the result, it is natural that we want to tell others about it. But you need to apply a lot of discipline here. In communicating what you've done and conclusions you made, we are simply not interested in the lines of code you wrote. Rather, we want to know what the telecommunications principle or process was that you were investigating, what you measured, how you interpreted the results and what you learned as a result.

Project Topics

1. **Aliasing**
   In lectures, we discussed quantisation accuracy and sampling rates when converting analogue signals to digital form. You had a lab on quantisation accuracy, but in this project you will investigate the sampling rate, including the appearance of aliased frequencies. Your aim is to use Matlab simulations to investigate resampling of audio or image signals (or some other signal, if you prefer), and the filtering necessary to avoid aliasing artefacts. Along the way, you will gain some experience with anti-aliasing filters. What do aliasing artefacts sound (or look) like? Can you show the effect in your presentation?

2. **Convolutional Coding**
   Many practical systems use convolutional codes (rather than block codes) for error correction. With convolutional codes, the coder operates on a continuous input sequence rather than on blocks of data. Write programs in MATLAB to demonstrate the operation of a convolutional coder and a convolutional decoder. Find information about the convolutional code used in some telecommunications equipment (e.g. digital television) and modify your program to implement that code. What variations can you try?

3. **CRC checks**
   We met polynomial codes, or CRCs, in lectures and you briefly used them in one of the laboratories. This project allows you to explore the capabilities of polynomial codes more fully. You should consider the variables that are important in error correction or detection and design an experiment that will allow you to present, graphically, the trade-offs involved. It might not be practical to vary all of the parameters and look at all the trade-offs, so you need to decide what is most important, what questions you want to answer, and design the experiments accordingly, using the Matlab built in polynomial coding functions as your principle tools. What is the trade-off when choosing one CRC over another (e.g. consider values of n and k for a particular probability of bit error), and why do you think long or short CRCs are used as they are?

4. **Interleaving**
   Interleaving is a method of spreading out bursts of errors so there is a better chance of correcting them using error correcting codes. In this project, you will use MATLAB to simulate and demonstrate how interleaving can be used to separate errors occurring in transmission before they reach the decoder. Your simulations will demonstrate the trade-offs between interleave depth and delay, and compare the two main categories of interleavers: block and convolutional. Interleavers work in conjunction with error correcting codes; which error correcting code will you choose for your demonstration and why? What error burst statistics will you use?

5. **Cache lookup algorithms**
   Whether ARP (in IPv4 networks) or NDP (in IPv6 networks) are used to obtain the addresses, a host or router stores the IP-MAC address mapping in a local cache. When a packet is to be transmitted, the host searches this table when a MAC address is needed to forward a packet to an Ethernet link.
   In this project you will investigate alternative lookup algorithms that can be used for such cache lookup operations. You will also choose three algorithms, and write a MATLAB program to compare their performance (you will need to think about the performance

criteria as well).
A useful book for the starting point is “Network Algorithmics” by G. Varghese. There is an online version of the book available through the Monash library.
(Note that the focus of this project is the efficiency of search algorithms, and not the use of the address information found.)

6. Image coding
The Discrete Cosine Transform (DCT) is a transform that has been used in image and video coding for many years - it is the basis of JPEG. In this project, you will use your own or standard test images, segment them into blocks of 2^n x 2^n pixels and convert them using the "dct2" transform in Matlab. While doing this, think about why the segmentation is performed; why not just transform the whole image as one big array? Once transformed (that is, “in the DCT domain”), you can then explore the effect of quantising the coefficients in the DCT domain - that is, representing them only approximately, or even zeroing those with small magnitudes. This would save many bits when transmitting the remaining coefficients to a decoder (that would then perform an inverse transform operation on the quantised coefficients it receives), but if you throw away too much information, you will see the effect on the pictures! Can you plot the distortion (measured as sum squared error) against the number of DCT coefficients that you throw away? What dependency do you see on “n”?

7. Video coding
When compressing video signals, one of the greatest savings in transmitted bits is achieved by estimating as accurately as possible each new video frame, based on previously transmitted frames. This is done by segmenting each new frame into blocks of 2^n x 2^n pixels and finding the best match for a block of pixels in the current frame, using a block of <u>nearby</u> pixels in a frame that has already been transmitted (most obviously, the previous frame). We then need send only the difference (the “prediction error”) between our estimate and the actual new frame pixel data (along with the motion vector). This method of coding is called motion compensation (MC). In this project, you will load a video sequence into Matlab and then segment each frame into blocks of 2^n x 2^n pixels. You could then perform MC on one or several frames, and could compare the size of the prediction error signal both with and without MC. (“Without MC” means you use the same pixel locations from the previous frame to estimate those in the current frame; i.e. the motion vector is zero.) You could then compare search strategies (e.g. exhaustive search, logarithmic search) to find the best match with some nearby part of the previous frame. Your analysis might consider such things as the processing complexity necessary for different search methods, the accuracy of each, the effect of the size (n) of the blocks used for matching, and the number and accuracy of motion vectors generated. (You are not expected to do the coding of the prediction error; your task is to focus on the performance and parameters of the MC.)

8. Audio coding
Audio coders such as MP3, Dolby Digital and AAC are all based on a characteristic of the human psychoauditory system called "Psychoacoustic masking". The existence of large magnitude frequency components in an audio signal means that we cannot hear, or cannot detect distortion in, other nearby frequencies. In this project, you will load a music file into Matlab, break it into time segments, take the FFT (Fast Fourier Transform) to obtain a frequency-domain representation, and explore what happens when you remove or coarsely quantise some of the lower level signal components before reconstructing the audio signal.

You are not expected to build a whole psychoacoustic encoder/decoder pair, but you should consider what aspects of this process you can explore, draw conclusions on and present. You could plot the error in the audio (e.g. sum squared error) against the number of coefficients you delete. Are there coefficients you can delete which cause no audible difference? A suitable reference text on this topic is “Digital Audio Coding and Standards”, by Marina Bosi and Richard Goldberg (Springer, 2003), which is available online from the Monash library.

9. **Packet queues in a router**
Queueing Theory is useful to analyse a vast number of processes involving flows of material items (like components in a factory, people in supermarkets, vehicles on roads,…) or virtual items like data packets as they pass through a network. In particular, a router includes a switch fabric that sends packets towards their destination, but packets may arrive in a bunch, and must sit in a buffer until they can be dealt with; hence the "queue". In this project, you will investigate the behaviour of queues under various arrival and processing conditions. Using Matlab, you should simulate the arrival of packets arriving at a switch (using a Poisson distributed random number generator to tell you how many packets arrive in each time slot), and also empty that queue using a similar Poisson distributed server process. How does the fullness of the buffer behave (particularly over the long term) as you vary the average arrival and server rates? What is the average delay experienced by a packet passing through this router/queue? Is the behaviour different if the server processes the same number of packets each time slot (i.e. the server rate is fixed instead of random)? You may like to explore other scenarios (such as multiple queues served by a single server, or multiple queues in series – modelling packets passing through multiple routers) in your investigation of queueing. You will do your own research to enable you to carry out this project, but a suitable starting point for queueing theory is Chapter 8 of "High Speed Networks and Internets: Performance and Quality of Service", by William Stallings, Prentice Hall, 2nd Edition 2002 (available in the library).

10. **Buffering and delays for compressed video**
When video data is compressed efficiently, there is a variable data rate generated for each video frame; the encoder will generate $E(i)$ bytes to represent frame i, but we might be using a constant transmission rate that transfers $T(i)=T$ bytes each video frame time. A buffer will be needed to allow these two processes to work together. Similarly, at the decoder, $E(i)$ bytes will be needed to reconstruct frame i and this many bytes will be read out of a buffer that is filled at a constant T bytes/frame. In this project, you will use data about bit rates generated for sample video sequences to investigate how much buffering is required in the delivery chain for encoder and decoder, and therefore what unavoidable end-to-end delay is involved (no matter how fast the processors are that are doing the encoding and decoding). By using “frame trace” data (that is, a record of how many bytes are generated for each frame in a compressed video sequence) and recognising that the frames are both generated at the encoder and consumed at the decoder at fixed time intervals, you can simulate the buffers necessary at either end of a transmission link using Matlab or Excel. These buffers will allow for variable rates to be generated/consumed, even though a constant rate might be transmitted over the network connection. If the buffers are ever allowed to empty, your decoder will be fed blank data; to make sure this doesn’t happen, your simulation should allow them to partly fill before you read anything out of them. How will you decide on the bit rate that you simulate between encoder and decoder? How big do the buffers need to be? What delay is involved?

Suitable frame trace data is available on the web (see http://www-tkn.ee.tu-berlin.de/research/trace/trace.html, or you may find an alternative source you prefer).

11. Equalisation in a communication receiver

Inter symbol interference (ISI) caused by multipath distorts the transmitted signal causing bit errors at the receiver. Traditionally, a time domain equaliser is used to combat distortion induced by the ISI before recovering data bits. In this project, you will equalise a time dispersed (affected by ISI), noisy, digitally modulated signal (eg., BPSK or QPSK) using Matlab's built-in function objects `lineareq` (a linear equalizer) and `dfe` (Decision Feedback Equaliser (DFE) – an example of a nonlinear equaliser). Consider at least two signal to noise ratio (SNR) conditions around 10dB to 20dB, and generate scatterplots of digital signals before and after linear equalisation and decision feedback equalisation. What are the advantages and disadvantages of using a DFE over a linear equaliser?

12. Analysis of Optimal Frame Size and Bit Error Rate Relationship on Lossy Links

Write a Matlab script that, through a graphical user interface panel, accepts a bit error rate BER range (minimum and maximum values) of an imperfect link (in which any packet with bit error is discarded), and a frame size range. It will then calculate normalised effective throughput (that is, data bits received successfully, as a proportion of total transmitted bits). This should be presented as a 3D-graph (X-axis: frame length, Y-axis: BER and Z-axis normalised effective throughput). (Hint: Your "frame" should simulate a real world frame (such as an Ethernet frame), taking account of the overhead involved, and you should consider the relationship between BER and lost data in this frame-based environment.) What can you learn about "optimum" frame sizes in this error-prone environment? Extend your study to take account of the extra overhead transmitted when there is retransmission to replace an errored packet; how big should the packets be to minimise the TOTAL throughput, and how does it depend on the BER? (Ignore the retransmission request traffic going back the other way.)

13. CDMA (Code Division Multiple Access)

In CDMA, each message bit is represented by a pseudorandom series of much shorter bits when transmitted, and we rely on the orthogonality of these "spreading codes" to enable us to separate the messages again at the receiver. Write a program which uses a Walsh-Hadamard (WH) code to spread a binary sequence to create a CDMA signal. The spreading sequence should be at least eight bits long. Demonstrate how a random data stream can be recovered by multiplying the received CDMA sequence with the correct WH code. Show how the orthogonality property of the code ensures that different transmissions can be recovered if power control and synchronisation is correct. You should demonstrate the operation of CDMA in a multi-user scenario, by using orthogonal spreading sequences to decode a required message while rejecting others. While you may wish to demonstrate the principle using short spreading codes, your simulation should use a spreading code that is at least 8 bits long for every input data bit. What is the effect of noise on the combined signal? What would be the effect of imperfect power control or imperfect synchronisation? You should consider not just integer bit shifts in your synchronisation simulation, but also shorter-term fractional bit offsets.

14. OFDM

Orthogonal frequency division multiplexing (OFDM) is a modulation scheme where many streams of data are transmitted in parallel on different orthogonal frequencies. Data sequences A_k and B_k are carried by the in-phase and quadrature components of the complex signal and represented by

$$x(t)=\sum_{k=0}^{N-1} A_k \cos(2\pi kt)+B_k \sin(2\pi kt) \quad 0<t<T$$

Where, T denotes the OFDM symbol period in the time domain. For digital television in Australia, $N = 8096$ and $T = 1ms.$ A_k and B_k for k = 0,1,..N-1 occupy independent sub-carriers and hence do not suffer from self-interference. This orthogonal condition across multiple sub carriers is used in the demodulation of A_k and B_k . To obtain A_k the received time domain waveform is multiplied with the cosine component of the kth subcarrier and is represented by

$$A_k=\int_0^T x(t) cos 2\pi kt \; dt$$

$$A_k=\int_0^T \left(A_0 \cos(2\pi 0t)+A_1 \cos(2\pi t)+A_2 \cos(2\pi 2t)....A_k \cos(2\pi kt)\right) cos\left(2\pi kt\right) \; dt$$

$$A_k=\int_0^T A_0 \cos(2\pi 0t) cos\left(2\pi kt\right) dt+\int_0^T A_1 \cos(2\pi t) cos\left(2\pi kt\right) dt....\int_0^T A_k \cos^2(2\pi kt) \; dt$$

It can be seen in the above demodulation process that all integrals over the symbol period are zero except for the last term involving $\cos^2(2\pi kt)$. Similarly, demodulation of B_k involves multiplication with sinusoidal component of the kth sub-carrier and integration over the symbol period which is given by

$$B_k=\int_0^T x(t)\sin 2\pi kt \; dt$$

Simulate the OFDM modulation in Matlab and plot the histogram of the amplitudes of the instantaneous signal values. Find information about another communication system using OFDM, adapt your program to use these values of N and T and repeat the plot. Plot the spectra and measure the Peak-to-Average power ratio of the transmitted signals and see how both of them change with N.

15. Bit synchronisation in radio systems

When a radio system commences sending some data (e.g. when a packet of data bits is transmitted over Wi-Fi), the receiver must be synchronised to start interpreting received bits as soon as the message begins. But transmission is asynchronous (there is no continuous transmission in between packets that would maintain synchronisation), so at the start of each message the receiver has to determine a) where the bit boundaries are, and b) when the actual message begins. This is achieved by transmitting a "training sequence" of a known bit pattern before the message payload. The receiver can try to correlate the received signal with the bit pattern it is expecting, to obtain its required timing information.

In this project, you will investigate the design of such training sequences. What are the desirable autocorrelation and cross-correlation features of such sequences? How long should a training sequence be, and what are the trade-offs when you make them longer or shorter? Perform a simulation using Matlab to investigate these things quantitatively. You do not need to modulate a carrier signal with the bit sequence, but you should experiment with what happens if the bit sequence is distorted or noisy. (Hint: One of the popular training sequences is the "Barker" family of bit sequences. This might be a good term to start searching for when you start your research.)

16. Project topic suggested by student and approved by staff

Or propose another topic and have it approved by Mike Biggar (Clayton) or Mohamed Hisham Jaward (Sunway). Before suggesting it, make sure it satisfies the following criteria:

- It is not the same (or close to) another project in the list above (which you wanted to do, but someone else got it first)
- It must involve some experimentation or simulation (e.g. Matlab)
- It must involve data collection and analysis
- You will still need to satisfy the other requirements of references, report length, etc.