University of Pennsylvania
Department of Electrical and Systems Engineering
Digital Audio Basics

ESE250 Spring 2012 Lab 5: Nyquist-Shannon Sampling Theorem Friday, February 10, 2012

For Lab Session: Tuesday, February 14, 2012 in Detkin Lab.
Due: Monday, February 20, 2012 by noon.

Collaboration: Work in lab in teams of 2. Perform individual writeups. See course collaboration policy in the Administrative Handout.

Objective: Appreciate and experience sampling rates; understand aliasing.

Prelab Requirements: Download and read this entire lab assignment especially the Motivation section and the Theory subsection of the Lab Writeup Guidelines section. **Please do the prelab questions in the Theory Section. These will be checked at the beginning of lab.** Remember to bring your headphones to lab with you. Optional but highly encouraged, encode a track from one of your music CDs into a `.wav` file sampled at 48KHz, 16bits.[1]

Deliverable:

- Names of all lab group members
- Answers to all lab questions (including prelab question)
- Your version of `generalSampling.vi`

Handin: All labs will be turned in electronically through the Penn Blackboard website. Go to the assignment submission link and follow the instructions. Please submit all your files as **one zip file**. Also, remember that your writeup should be a **PDF file**.

Exit Ticket: Play for your TA an under-sampled version of your voice recording.

[1] Make sure you encode it from a CD, re-encoding an existing `.mp3` into a `.wav` will not be useful for the lab.

Motivation

When sampling, we must take care to ensure that we are sampling at the right frequency to prevent aliasing (a phenomenon that occurs when you sample at a lower frequency than is needed). In this lab you will investigate the relationship between the sampling rate and the frequency content of the sound to be sampled. You will look at aliasing, how it affects sampled waves, and how to prevent aliasing.

Lab Procedure

Sampling Pure Tones

In class we did a group exercise where our sampling rate was not high enough to capture the true form of the signal. In this part of the lab you will be able to experiment with different frequencies and sampling rates and see, first hand, why aliasing happens. To better exhibit the effects, the original signal will only be pure tones.

- Make sure you have a copy of lab5.zip unzipped and load `toneSampling.vi`

Figure 1 shows the front panel for `toneSampling.vi`. On the left you have the controls that let you adjust the frequency of the signal we want to capture (Frequency of the Signal to be Acquired) as well as the Sampling Frequency at which you will sample points from the original signal. You can also play the original signal and the sampled signal and see what is the ratio of sampling frequency to signal frequency.

On the right you will find three graphs. The top graph shows the time domain of the original signal (in thin blue lines) and the sampling points (black points) and linearly interpolated curve they form (in thick black dotted lines). The bottom left graph shows the same time domain view but over a much shorter time period so that you can more easily see the details of what is happening when you go to high frequencies. The last graph, lower right, shows the frequency response of the original signal (blue spike) as well as the perceived frequency of the sampled signal (black spike).

- Run the VI and see and hear how the signals change as you change both sliders

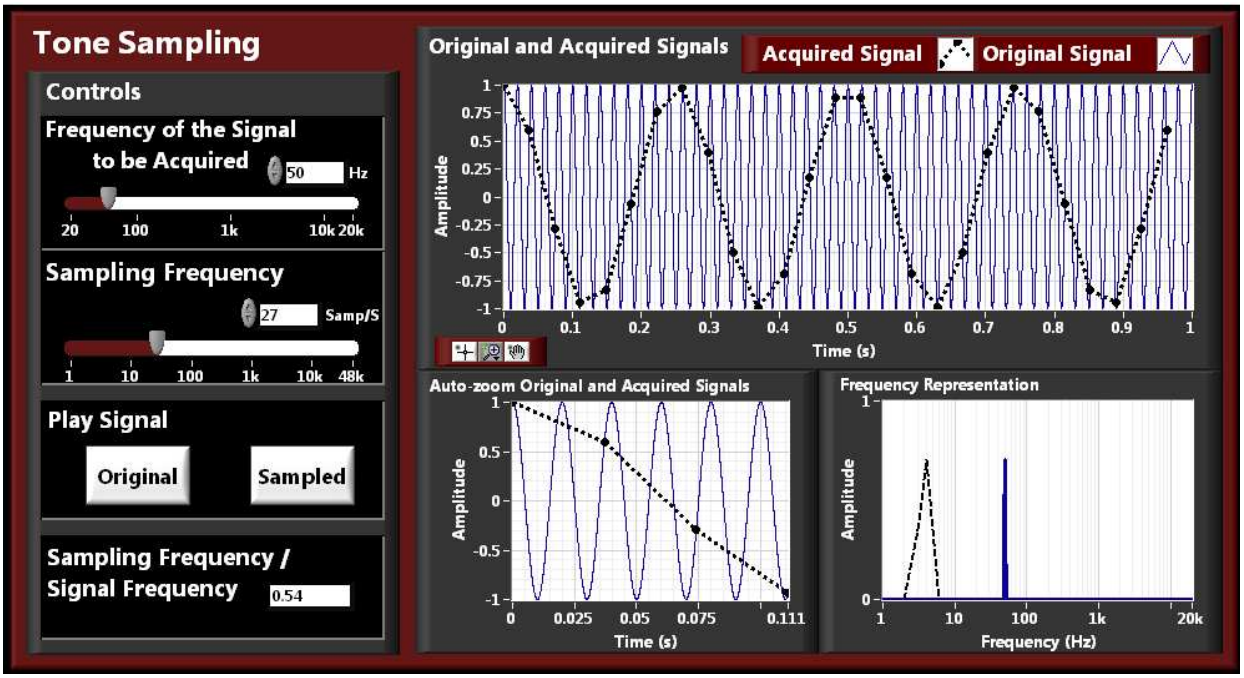


Figure 1: Sampling pure tones

FOR QUESTIONS:

1. Set the frequency to 440Hz.
2. Record the minimum sampling rate at which you see only the input frequency on the frequency representation graph.
3. Find the ratio of the sampling rate to the original frequency.
4. Change the frequency and repeat steps 2 and 3 (try some higher and some lower frequencies).
5. Fix your frequency, f.
6. Adjust your sampling rate to

   Less than f
   Equal to f
   Greater than f but less than f times the ratio found in previously.
   Greater than f times the ratio you found previously.

7. Observe what happens to the look and sound of the wave when you adjust the sampling frequency.

Aliasing Noise

Aliasing noise happens when the original signal has information at higher frequencies than the sampling frequency. These high frequencies are aliased to low frequencies and cause unwanted interference or noise. Getting rid of this noise requires applying an anti-aliasing filter before sampling the signal. Since the aliasing noise comes from high frequencies, the anti-aliasing filter that we use is a *low-pass filter*.

An ideal low-pass filter allows all frequencies below a specified frequency to pass and completely removes any frequencies above the specified frequency. This allows us to get rid of the part of the signal that would otherwise be aliased and produce noise.

- Load `AliasingNoise.vi` and run it.

With `AliasingNoise.vi` you will be able to simulate the effect of noise aliasing into the signal. Figure 2 shows the front panel of this VI. As with the previous VI, the left side is the controls and the right side has graphs. The controls allow you to specify the frequencies of both the signal and the noise, set the sampling rate and define the cutoff frequency for the low-pass filter. You can also listen to the filtered and unfiltered signals.

The three rightmost graphs are frequency domain graphs of the filtered and unfiltered signals as well as the difference between the two, showing which frequencies were removed by the filter. The Graph Key in the middle of the VI explains what each frequency marking on the graph is. Green and black dotted lines show the frequencies for the filter and sampling rate. The blue lines represent the signal, bright blue for the specified frequency in the controls and light blue if there is an aliased frequency. The frequency of the noise is marked by a red line, again bright red for the specified frequency in the controls and light red if there is aliasing.

The final two graphs, Filtered Time and Unfiltered Time, show the filtered and unfiltered signals in the time domain. You can zoom in and out of these graphs to examine how the signals change as you change the controls.

When you first run the VI, it will show a signal of 400Hz and noise at 3,300Hz. You will also see the noise aliasing to 300Hz and the filter, at 600Hz removing the noise and aliased noise and leaving the signal intact.

As you change the sliders in the control, you may notice that the Error lights light up on the Signal and Noise controls, you should ignore any effects you see when this happens and change the sliders by 1Hz to get rid of these errors. The error is due to an internal LabView VI.

FOR QUESTIONS: Leaving the default values:

1. Observe how the filtered and unfiltered signals differ, both visually and audibly.
2. Lower the noise frequency until it no longer aliases. Record the frequency.

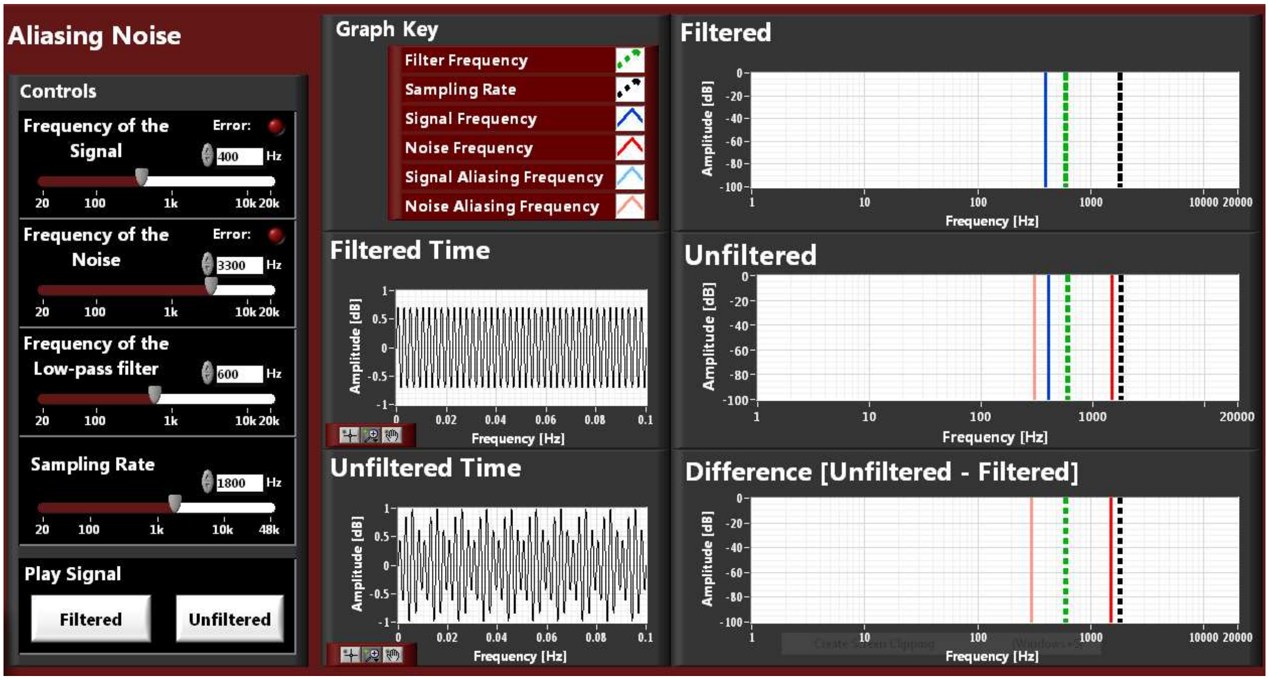


Figure 2: Aliasing noise

3. Keeping the noise to the frequency discovered in the previous question, adjust the frequency of the low-pass filter to the maximum frequency you can set the filter to before you see noise in the Filtered graphs. Record that frequency

4. Rounding to the nearest whole number, find the ratio of the sampling rate to the filter frequency

Pick new values for the four sliders so that they meet the following requirements:

- $f_{signal} < f_{low-pass} < f_{noise}$
- $f_{sampingRate} > 2 \times f_{signal}$

Write down the new values and repeat steps 2 to 4 for these new values.

Signal Reconstruction

From lecture and the last two sections you know that if you set the sampling frequency correctly and filter your signal you can perfectly capture your signal up to the Nyquist frequency. Eventually you will want to reconstruct the analog signal from the sampled points. However, as you will see in this section, simply "connecting the dots" will not produce the original signal faithfully.

- Load `reconstructSignal.vi` and run it.

This VI (Figure 3) shows two different ways of reconstructing the sampled signal. It lets you specify a frequency of the original signal as well as the sampling frequency. Both graphs show the sampled points. The top graph shows the signal obtained by joining the sampled points with a straight line. The bottom graph shows what the correct reconstruction looks like. It does this by first converting the time domain samples into the frequency domain using a DFT and then using the frequencies discovered, it reconstructs the time signal by combining the frequencies (similar to what you did to simulate instruments in lab4).

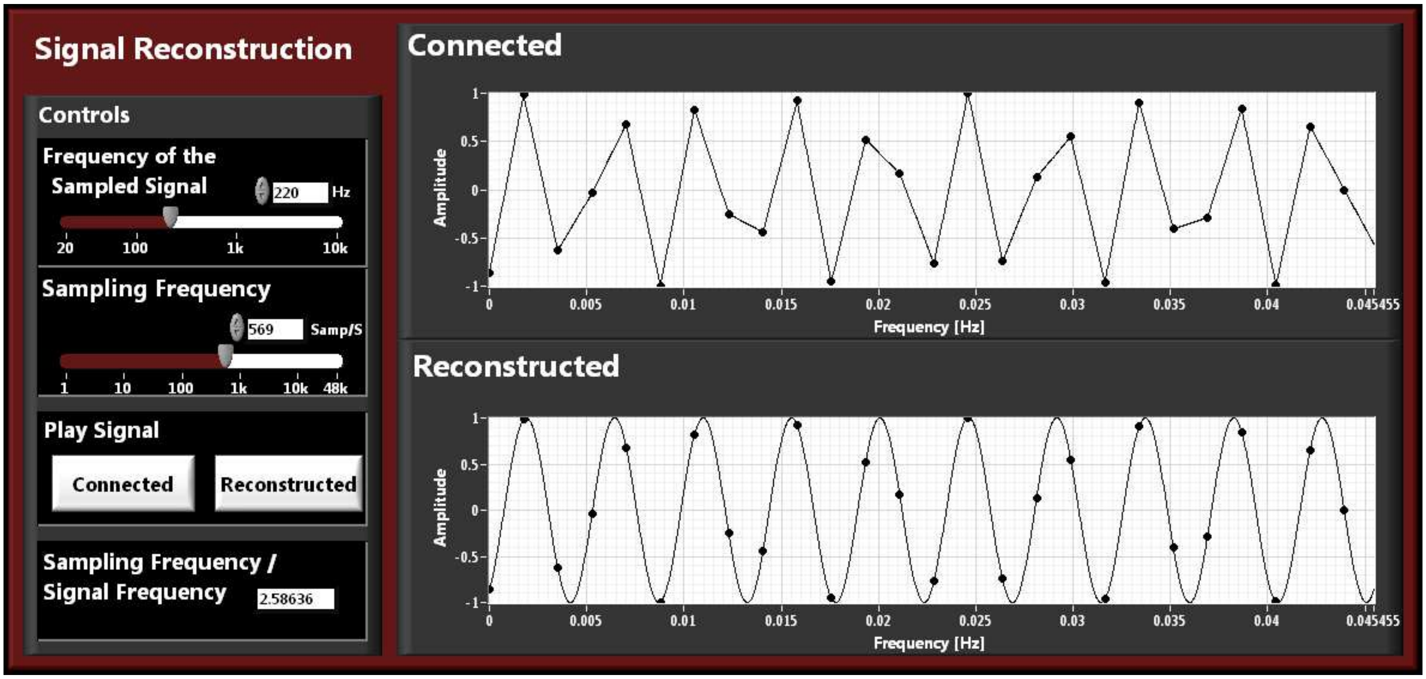


Figure 3: Signal reconstruction

Set the frequency and sampling frequency to the ratio you learned in class and discovered in section **Sampling Pure Tones** of this lab.

FOR QUESTIONS:

- Observe how the two types of reconstructions look and sound.
- Find the sampling rate you need to make the “connect the dots” wave sound and look like the reconstructed wave when frequency is set to 440Hz.
- Record the ratio between this sampling rate and the frequency.
- Vary the frequency and find the ratio a number of different frequencies.

Sampling Arbitrary Signals

Now that you understand what is required for successful sampling and signal reconstruction, you will build a simple VI that lets you sample and reconstruct arbitrary signals stored in a `.wav` file. If you brought an encoded piece of music, make sure you have it handy, if not, download one of the `.wav` files from Blackboard.

We will also make a voice recording that you will use as another sample signal.

- Load `recorder.vi` but don't run it.

This VI lets you set a recording length and a file path where you want to store your recordings. When you run it, it will record from the microphone and display the recorded signal on the graph.

- Connect the microphone to the computer.
- As you create recordings, if you find some of the waveforms are being clipped lower the volume setting on the microphone.
- Record one or two conversations between you and your lab partner.

The VI you will build will allow you to choose a `.wav` file, the sampling and anti-aliasing filter frequencies, the reconstruction type (connect or reconstruct) and whether to apply the filter or not. It will also play the sampled signal and display it on a graph. The final block diagram for the VI will look similar to figure 4.

- Open a blank VI
- Right click on the front panel and under the **Buttons & Switches** tab choose a switch that will be used to turn the filter on and off. Place it anywhere on your front panel.
- Again, right click on the front panel and place a **Graph** from the **Graph Indicators** tab.
- Switch to the block diagram `ctrl+E`.
- Right click on the block diagram to bring up the functions menu, choose search.
- Search for "Sound File Read Simple.vi" and click and drag it to the block diagram.
- Bring up the functions menu and under **Express** → **Signal Analysis** choose the Filter.
- The filter configuration dialog will come up. Leave the default values but set *Order* to 10.
- Connect the **data** terminal of the Sound File Read Simple.vi to the Signal terminal of the filter. (Open the context help to see which terminal is which, `ctrl+H`).

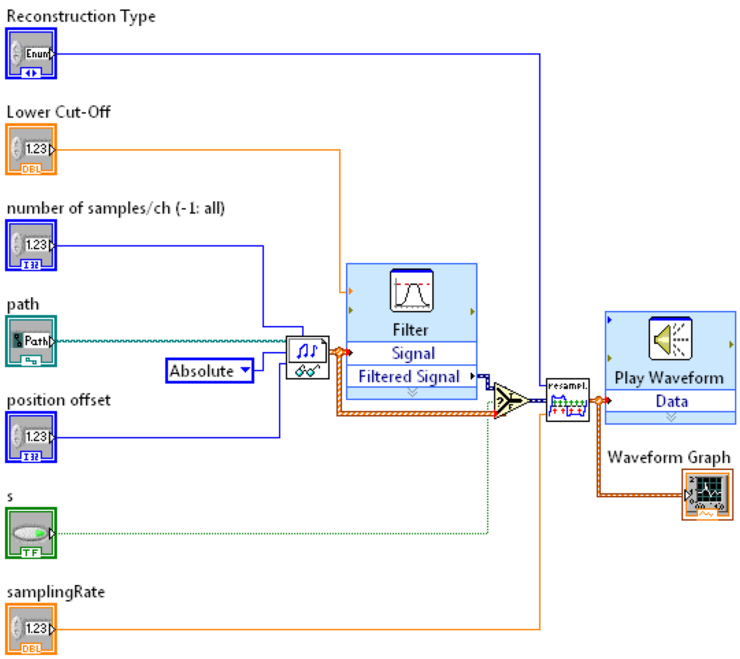


Figure 4: Block diagram

- In the functions menu, choose search and look for the Comparison submenu. Within Comparison, choose the Select block and drag it to your block diagram. The Select block lets you choose between two values based on the state of a switch.
- Connect your switch to the **s** terminal of the Select block (the middle green terminal)
- Connect the **Filtered Signal** terminal of the filter to the **t** terminal of the Select block.
- To the **f** terminal of the Select block, wire the **data** terminal of the Sound File Read Simple.vi block.
- From the functions menu choose **Select a VI...** and load `resampleSignal.vi` from the files you downloaded for the lab.
- Connect the Select block to the **Filtered Signal** terminal of the resampleSignal subVI.
- Select the Play Waveform... subVI from the **Express** → **Output** tabs in the functions menu.
- Leave the default values in the configuration dialog and choose OK.
- Connect the output of the resampleSignal subVI to the Waveform Graph and to the **Data** terminal of the Play Waveform subVI.
- Create controls for the following terminals by right clicking on them and choosing **Create** → **Control**

- Reconstruct Type terminal from the resampleSignal.vi. Lets you choose how to reconstruct the signal
- samplingRate from the resampleSiganl.vi. Lets you set the sampling rate
- Lower Cut-Off from the Filter subVI. Lets you set the low-pass filter cutoff frequency
- Path from the Sound File Read Simple subVI. Lets you specify the `.wav` file to use.
- Number of samples/ch from the Sound File Read Simple subVI. Lets you specify the length of the file to read in terms of samples. Adjust this so that you are playing between 15 and 30 seconds.
- Position offset from the Sound File Read Simple subVI. Lets you specify at which sample to start reading, if you set it past the end of the song LabView may complain.

- Create a constant for the position mode terminal of the Sound File Read Simple and set it to Absolute.
- You may cleanup your VI by pressing `ctrl+U`
- Switch to the front panel and arrange the controls any way you want.

You are now ready to sample arbitrary signals. When answering the following questions you may choose any sampling rate-filter pair but we suggest you try some of these sampling rates: 2kHz, 4kHz, 8kHz, 16kHz, 32kHz and 44.1kHz as well as these filter settings: No filter and filter set to just under half the sampling rate (Note that if you set the filter equal to or greater than half the sampling rate, LabView will complain and your VI will not run)

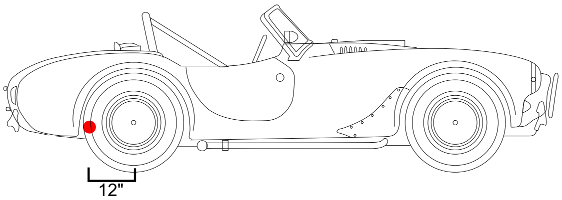
FOR QUESTIONS:

- Play one of your voice recordings at a high sample rate and adequate filter settings. Find the lowest filter and sampling rate at which it no longer sounds clear.
- Adjust sample rate and filter to the lowest level at which the voice recording still sounded OK. Play a piece of music with these settings.
- At the sample rate and filter of the previous question, play for your TA your voice recording (required for **Exit Ticket**).
- Raise the sampling rate and filter settings so that the music sounds clear.

Lab Writeup Guidelines

Theory

Prelab Question: Consider a movie shot at 30 frames per second. The scene being shot is of a getaway car accelerating from a stopped position. The rear hubcap on the side of the car facing the camera has a single red spot 12" from the axle.



When the car reaches what speed will the red dot begin to appear to stay still relative to the actual motion of the wheel? At what speed(s) will it appear to rotate backwards?

Analysis

Sampling Pure Tones

Question 1:

- For your 440Hz wave, what is the minimum sampling rate at which you no longer experience aliasing?
- What is the ratio of the sampling rate to the original frequency?
- Does this ratio stay the same when you go to a higher/lower original frequency and adjust the sampling rate to the minimum sampling rate at which you no longer experience aliasing for the higher/lower frequency?

Question 2: Keeping the acquired signal's frequency f fixed, describe visually and audibly what happens when the sampling rate sf is:

Less than f
Equal to f
Grater than f but less than f times the ratio found in Question 1
Greater than f times the ratio you found in Question 1.

Aliasing Noise

Question 3: Describe how the filtered and unfiltered signals differ, both visually and audibly.

Question 4: To what frequency do you have to lower the noise until it no longer aliases?

Question 5: Keeping the noise to the frequency discovered in the previous question, what was the maximum frequency that you had to adjust the frequency of the low-pass filter before you saw noise in the Filtered graphs?

Question 6: Rounding to the nearest whole number, what was the ratio of the sampling rate to the filter frequency?

Question 7: Will the ratio of sampling frequency to filter frequency be the same for any of the new values you pick and adjusted for? Explain why or why not.

Signal Reconstruction

Question 8: Describe how the two types of reconstructions look and sound?

Question 9: What is the sampling rate you need to make the "connect the dots" wave sound and look like the reconstructed wave when frequency is set to 440Hz? What is ratio between this sampling rate and the frequency? Is this ratio consistent as you change the frequency?

Sampling Arbitrary Signals

Question 10: For your voice recordings played at a high sample rate and adequate filter settings, what is the lowest filter and sampling rate at which it no longer sounds clear?

Question 11: For the piece played with voice settings, briefly describe how the voice and music differ? Why do you think they differ so?

Question 12: To what value do you have to raise the sampling rate and filter settings so that the music sounds clear?

Conclusion

Question 13: Given what you discovered in the previous questions, what can you conclude about how sound is processed on the cellphone (Think back to what it sounds like when you speak with someone and when you've been put on hold and hear music).

Further Thought

This section is **not** *part of your required assignment. Along with each week, we will offer directions and questions for further thought. Due to the nature of this course, we can only begin to glimpse the depth and richness of each of the topic areas. These questions offer some headings to contemplate further depth. These will often be open-ended questions.*

Imagine that 20 years from now we finally find some Martians, but it turns out they have a higher frequency of hearing and sound production than we do. Perhaps they use frequencies up to 200KHz compared to our meager 22 KHz hearing. (On Earth, marine mammals including dolphins communicate with frequencies up to 150–200KHz.) The Martians lack digital communications technology but do produce many other advanced technologies for trade. As such, the Martian market looks like a good untapped market for expansion.

1. The martians find that our communication equipment (cell phones) doesn't work for them. Why would that be?

2. How would we produce digital equipment they could use?

3. Perhaps some very dedicated Martians can limit their speech to a range we can hear, but that's likely to require great skill, training, and difficulty. To be polite to the Martians, it would be appropriate to communicate with them directly in their preferred range of speech and hearing. How could we communicate with them? What kinds of digital sound processing would help us bridge the gap? While they are visiting Earth, perhaps their cell phones need to allow them to speak with both humans and other Martians.