FREQUENTLY ASKED QUESTIONS

QUESTION

Q9: Expanding aperture is a technique in ultrasound beamforming where the objects very close to the aperture are imaged by a lower number of aperture elements. That is, as the depth gets higher, the number of active aperture elements becomes higher or the aperture is “expanded”. To show that, write a Matlab code that simulates the field of an ultrasound linear array of 32 elements with element width of 0.5 mm and kerf of 0.1 at a frequency of 5 MHz. Compare the field at 1 cm, 2 cm, 3 cm, and 4 cm with different number of array elements to show this. Is there a rule that you can derive from your observation to tell us when to turn on array elements?

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if we make imaging at 1cm number of active elements that we actualy uses will be small that means the width at the spatial domain will be small give me wide band signal at freq bad resolution . if depth increase number of active elements increase and give a narrow band signal at freqand good resolution.

i think i want to take foureier transform for 8 rect elements and FFT for 16 elements and for 32 elements and compare the field at each case

Note:I regarded that we use a lens so that the field for near and far calculted using fraunhover (FFT for the aperature)

ANSWER

You cannot use a lens for this purpose. In fact, expanding aperture was invented because when you are too close to the aperture the elements on the side will require very large delays compared to the middle ones and such delays cannot be realized in practice. Just look at it from the points of view that you are effectively combining the fields of the individual elements. When the point of interest in making 0 degrees with one element and nearly 90 deg with another, the field from that other one will have no effect on your aperture. The purpose of this problem is to make you realize that this is in fact that case by doing such simulations. You will find that the field is much better when you have smaller number of elements when you are imaging at very low depth. Since there is not much time left, I suggest that you just take this as a fact since such simulations might take some time.

QUESTION

check this question related to Frequency vs. Time trade off

if the depth increase so Tprf increase and prf decrease so the max. velocity will be decrease because the max. shift that will be detected is prf/2 (to avoid aliasing)but the resolution at the frequency will be good , but the resolution by definition is T/2 which T is the over all pulse width so the resolution at the time will be bad the question now does this conclusion true? and what is the effect of the bad time resolution i draw the power spectrum the power related to the shifted frequency i cant understand the effect of such this problem with time resolution.

ANSWER

All what you said is correct but the conclusion is not. The axial resolution is important for B-mode only. In PW-Doppler, you get the spectrogram of a sample volume with size around 1cm, not a single pixel. That is, you basically look at much bigger points in the Doppler mode. In practice, the excitations used for PW-Doppler are typically 8-16 times as long as those used for B-mode.

On another note, the “time” resolution I was talking about in the PW-Doppler mode is not the “axial” resolution that is equal to pulse length/2. In order to compute the spectrogram, one uses a number of pulses with Tprf in between, say 32 or 64 points. This means that a single line in the spectrogram will represent what happens in 32Tprf or 64Tprf. If the blood flow changes during such period, then the spectrogram will not be able to “resolve” such variations in time. That is why a window of smaller number of points (here 32) would give you a better time resolution than that of the one with more points (here 64). The frequency resolution will be exactly the opposite of that where the window with more points will have better frequency resolution.

QUESTION

I have some questions about Doppler ultrasound...

1.what is meant by **Frequency vs. Time Resolution Trade-Off and Time-Bandwidth product**?

2.what is the advantages of PW Doppler over CW Doppler?

3.I think you said before that about the Doppler, the model of transmitting a continuous wave is not applicable, i didn't understand this part. Need some clarification please.

ANSWER

1. Refer to page 21 of the Doppler lecture presentation to see what is meant. When you use a small sized window, you get a coarse resolution in the frequency domain, and vice versa. At the same time, having a small window allows you to have a better resolution in the time domain (which means you can follow changes faster). This is a consequence of eh Fourier transform property that the multiplication of the bandwidth in both time and frequency domains will be a constant. This means that you cannot improve the resolution in both domains simultaneously and you have to work on a trade-off to reach an acceptable compromise that works well for your imaging situation.

2. PW Doppler offers control over spatial localization of the sample volume. You cannot change the sample volume size or depth in CW Doppler.

3. In CW Doppler, the model is indeed continuous wave. In PW Doppler, you cannot do that because you rely on a short excitation to allow spatial localization (if you send continuously, you will never know where the signal is coming from, and remember that the axial resolution is the pulse length/2 and hence you will have no spatial localization at all with continuous excitation).

QUESTION

- In apodization:

1) How should we use the central frequency in the problem? just scaling in the x- direction by (Lamda*Z),right?
2) Sampling in created aperture, does it have any effect? (NONE, but better resolution in the field)
3) which are internal sincs and where are their side lobes? the envelope sinc is drawn in one of the images.
4) applying different windows affected the amplitude of the main lobe, why?

aliasing in the comb happens (element spacing Vs lamda/2), but we can't specify the grating lobes!

- We have ambiguity around the single frequency Vs band of frequencies, shall we simulate such other frequencies to see useful results?
- The Hadamard matrix: (code attached)

Running the code several times gives random results; the SNR sometimes increases, sometimes decreases.. did we apply the concept correctly?

- In Harmonic imaging: we just use the band of frequencies around 2fo (in concept)

In practice, we get the fft2 of the aperture, scale it at two frequencies (fo and 2fo): x-axis * Lamda*Z , and with the higher frequency (2fo) we have a narrower main lobe, right?

ANSWER

Apodization:

1) Yes.

2) I am not sure what you mean by “created aperture”. Sampling should only present suitable but finite number of points to see the field without missing major changes.

3) The plots seem to have a very poor resolution in the main lobe/side lobes while showing too many grating lobes. You should display the main lobe and 1-2 side lobes only since this should be enough to get the point. I am not sure I understand what you mean by “internal sinc”.

4) Actually, the magnitude of the main lobe should be the same if you keep the area under the window the same between all windows. Otherwise, since it is equal to the DC component of the window, it will vary with each window. Such multiplication factor is of no importance in the process since we can always multiply by a compensating factor. What we look for is the change in the side lobe magnitude (main reason for using a window) and its associated side effect of main lobe width increase.

I am not sure what aliasing you are talking about. There should be no aliasing in beamforming or apodization. Grating lobes are the peaks that you are seeing in your results, just compare the period to that of the grating lobe period to come to this conclusion.

Single vs. frequency band:

In order to simulate realistic ultrasound fields, you should do field calculation for multiple frequency values then do a weighted average with the same factors as these frequency components in the ultrasound excitation. To make it closer to you, just average the magnitudes of the field from 3-4 frequencies in the range of the ultrasound excitation.

Hadamard:

Why did you invert H with knowledge that it was chosen because its inverse is its transpose?

You should use independent signals, not the same signals. Otherwise, you will have no meaningful problem.

When you do simulations with random numbers, it is not correct to run the process once and look at the results. Every time you will get a different results of course due to the random nature of the simulation. If you want to get a useful result, you should run many times and average the results. This is not a part of the course though and you should have taken that in stochastic processes courses. Just understand the theory and you should be fine.

Harmonic imaging:
No. You compute the aperture with a particular frequency fo and then repeat the computation for double that, i.e., 2fo, then compare the two. It is not possible for you to simulate the nonlinear effects that results in getting back 2fo when the excitation contained only fo. You can also look at the theory to know what will happen to the main lobe and side lobes (through lambda).

QUESTION
when using frenales and fraunhover and exact solution?
at the problem of expanding aperture I cant see any effect for the depth when using fraunhover?! and you told us if the lesn used we can compute the field for the far and near field using faunhover, I know that the lens uses to make focusing at y direction and the focusing at x direction will be electronic focusing

ANSWER
It is up to you to decide which you prefer to use according to the conditions of the problem. Just make sure that the conditions of each is satisfied. The effect of depth appears in the "z" term in the equation. You replace fx for example with x/lambda.z , and when you change z you will get a scaled version of the same field shape (that is, it will remain Sinc for example but with a wider main lobe for a higher z). Yes you are correct about focusing techniques in x and y.

Question: ANSWER in red:

1) In the CW Doppler,, we said in order to get the spectrogram ,we stand at a time t and construct a window[this window might be a 128 point or so] centred at t and then get the FT the output spectrum is the frequency points plotted at a specific time t,, and we said that we can have more than one frequency corresponding to a distribution in velocities? Is that right?
   But in the PW Doppler in order to get a single window, or a single point in time in the spectrogram what do we do exactly?
   Do we draw the spectrogram for the single range of interest specified by the doctor or what?

CW Doppler is the same as PW Doppler as far as processing for spectrogram with the advantage that it gets the continuous domain Doppler shift then samples it. So, with CW Doppler, you do not have any aliasing problem. In PW Doppler, you specify a particular sample volume and so, you can localize the volume for Doppler analysis. CW Doppler has a fixed sample volume that you cannot see (remember the example of the fetal Doppler we talked about in the lecture).

What i got in the lecture is that the multiple pulses we send say 128 pulse corresponds to the window we defined in the CW Doppler and that the 128 periodic pulses gets us a single line in the spectrogram? Is that right?

Yes

Also i want to ask something,, the number of pulses i take to get a single line, won't that affect my ability in tracking the change in velocities?

Refer to the theoretical analysis of the PW Doppler we did in the lecture to see the effect of the pulse length. Another issue is that it should make your SNR better.

2) How do we obtain the shifts shown in the figure below by the red arrows? By the 128 excitation pulses responses or what? I noticed that the shifts are not equal, why?
   Also,, where is the Doppler shift, are we only concerned with the shifts around the central frequency ,i.e between the first impulse and the second or what?[i mean the pointed by the yellow arrow is our Doppler shift]

Refer again to your theoretical PW Doppler model. Your excitation will eventually amount to a series of pulses at certain frequencies. When Doppler shift occurs, it affects all of these pulses independently. From the Doppler equation, and since each pulse corresponds to a single frequency, you will have a different Doppler shift for each. You usually band-pass filter to get only the one in the middle (because it has the highest SNR).

3) Concerning slide in page 25, the on entitled **PW Doppler 2D Signal Model, i can't get it at all,,** for what exactly are we getting the 2D FT?? And what does the 2 D graph signifies?

I did not give that part much effort in the lectures and I told you that you can skip it.

4) You said in the lecture that we remove the clutter using a high pass filter,, i can't get this in the PW Doppler since by definition the clutter is the same as the sent signal but with time shift that's all,, here our sent signal has a 1 Mhz bandwidth,,meaning that the received clutter has the same bandwidth?? Right or wrong?

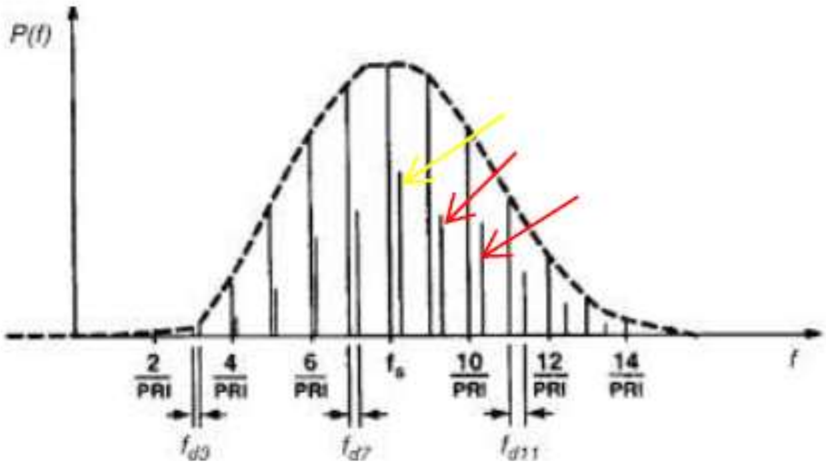
This is not correct. You are not filtering the returned signal. You are filtering the signal across 128 excitations for example (one point representing the sample volume in each line).

   Also, you said that we can get better clutter separation by increasing the window size? What is the relation? Do you mean that by this we reduce the width of the since at the central frequency???

The more points you have, the higher the filter order can get. If you have for example a 32 point window, you cannot use a filter with 32 taps and you usually use a filter order of something like ¼ of the number of points to get something useful.

5) I think in general, i have a problem in how we construct the spectrogram in the PW Doppler and how we can get the frequency shift that are embedded in the sampled 1 MHz bandwidth? So please explain it to me briefly

Each line in the spectrogram is a windowed FFT of the signal. You shift such window by an amount that suits the refresh rate of the spectrogram display (just like how you would display an ECG signal). The spectral magnitude squared is displayed as brightness in the spectrogram with each windowed FFT displayed as a vertical line (representing the frequency of velocity axis). New vertical lines are displayed side by side at consecutive locations on the x axis (representing time).



QUESTION

In Doppler, you said in slide 24 and 27 that the FT is periodic , from where does this periodicity comind from?

ANSWER

In PW Doppler, the input time signal is sampled. Consequently, the frequency domain will be periodic from the basic Fourier transform properties.

1-at the slides we have an example to calculate the typical tissue attenuation,example 10 cm penetration @ 5 MHz - 25 dB one way I cant understand!

ANSWER: (0.5 db/cm/MHz) x 10 x 5 = 25

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2-3-D I imaging constrains volume update rate :
Example:what is the meaning of the following?
60°x 60°x 12 cm pyramid volume?
1°beam spacing ⇒3600 beams?
12 cm x 2 / 1540 m/s = 160 μsec per beam?
1.7 volumes / sec?

ANSWER: 60 line (width) x 60 line (elevation) x (time for one line = 13 us x 12)= 576 ms to collect one volume. Then Volume rate is the inverse of that.

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3-delay calculations at steering for linear and convex array
I was trying to calculate the delay at the linear steering using the law available at the lecture but the results not reasonable example :lnegth between the second element and the wanted depth= (rX)where r is the depth and x is the distance between the two elements the resulted value very small

ANSWER: The formula is correct and you can derive it from the geometry. The delay values SHOULD be in the order of nanoseconds. Usually, the maximum delay is of the order of less than 1 microsecond.

QUESTION

I just want ask about the color flow imaging, i find difficulty to understand the slides Data acquisition in CFI (34) and the next two slides. Do you have extra material to read from !

ANSWER: The CFM is based on the correlation methods for velocity estimation. You can read the material posted for the Doppler lectures to learn more.

Also, you mentioned on the website, that we have to understand the material **deeply,** i don't understand what level of depth should i have?

ANSWER: To understand what I said in the lectures should be deep enough.

QUESTION in blue - ANSWER in Red

Trade-off between spatial resolution and frequency resolution in Doppler:

- In PW Doppler, what happens if the pulse length is decreased (Tg) ? Isn't the frequency resolution dependent only on (PRI * number of pulses) [outer window width containing the 128,64,.. pulses].. so what is the trade-off between spatial and frequency resolution in this case? I can only see it if we use the same pulse excitation for both B-mode and D-mode.

ANSWER: Mainly SNR not resolution. However, SNR is very important for Doppler applications given its small signal.

- In the optimal receiver, the pulse length affects both spatial and frequency resolution, (spatial resolution to determine where the shift happened) since we use matched filter detection of the received signal rather than frequency analysis of the Doppler signal we get from the received signal. Right?

ANSWER: Correct.

In CFI

- "Displays a color coded map of *axial* velocities" does this mean we dont give angle to the machine?

ANSWER: What you get is the axial component of the flow regardless of where the flow is going. The user can tell the system the angle of the vessel to try and get the full in-plane velocity but what is actually measured in the axial not the whole vector. Refer to Kadah and Tewfik, Engineering in Medicine and Biology Magazine, 1996 (available for download on the Publications section of my web site) for more explanation (this paper is not included in exam of course).

- In triplex mode, the CFI also is for axial velocities, indep of the angle given to the machine by the user??

ANSWER: Yes. This is true for all modes.

- Don't understand the Interleaved Packet acquisition. Is it the same as figure # 2 in previous slide? if so, is it bec: "it is important to avoid aliasing bec mean velocities are found, and aliasing would corrupt the data, so we need to increase the PRF"

ANSWER: All these methods attempt to strike a delicate balance between getting the max PRF while avoiding any blocking in the image due to having information for different parts acquired at long different times. So, there is no one method that can be considered optimal and each method has its pros and cons.

- What is the settling time for clutter filter?

ANSWER: It is the impulse response length, which is the order of the filter.

Phase aberration delay computations:

- what is the difference between making correlation in time domain and frequency domain? Is it cost of computations? what about performance?

ANSWER: Theoretically, you are computing the same thing. So, there should theoretically be no difference in performance. However, different methods attempt to make such computation more efficient by utilizing properties in one of these domains and/or making certain approximations. Different performance may result in this case depending on the kind of approximations involved.

Clutter filter:

- Can we remove clutter only in the digital domain?? What # of bits ADC is used then?

ANSWER: It is possible but the number of bits should be at least 12-14 bits (usual ultrasound sampling for B-mode needs only 8 bits).

QUESTION

i ask about how to estimate Doppler spectrogram by using fourier transform because i missed lecture and i can not imagine the solution that you explained from hearing the lecture

and the solution of the shape(c) in assignment one annular transducer

ANSWER: (Spectrogram: answered above and repeated here) Each line in the spectrogram is a windowed FFT of the signal. You shift such window by an amount that suits the refresh rate of the spectrogram display (just like how you would display an ECG signal). The spectral magnitude squared is displayed as brightness in the spectrogram with each windowed FFT displayed as a vertical line (representing the frequency of velocity axis). New vertical lines are displayed side by side at consecutive locations on the x axis (representing time).

As to the computation of the field, just decompose it in terms of rings. Each ring can be expressed as two circles subtracted (i.e., Circ() – Circ()). You know the transform of the circ from the lecture and you can then draw the field by Matlab.

QUESTION:

When we should use the exact, Fresnel and Fraunhaufer approximation when calculating the field diffraction pattern at any depth?

ANSWER: It is up to you to decide which you prefer to use according to the conditions of the problem. Just make sure that the conditions of each is satisfied. The effect of depth appears in the "z" term in the equation. You replace fx for example with x/lambda.z , and when you change z you will get a scaled version of the same field shape (that is, it will remain Sinc for example but with a wider main lobe for a higher z). Yes you are correct about focusing techniques in x and y.

Regarding the exact solution, we just will use the transfer function and use the frequency domain to calculate the fourier transform of the field, but when we inverse it to obtain the field, we can't make the integration over the complex functions. So, is the required to just use the transfer function to get the Fourier of the field.

ANSWER: No. What you want the field as a spatial distribution as a function of x and y. You use the method that makes it easier for you to do that. People discussed approximations like the Fresnel or Fraunhofer formulae to make it easier to do that. So, use what allows you to compute the field easier provided that you are not breaking any assumptions.

Fro the condition of the fresnel nd fraunhaufer, (found at page 43 in the Book) what are exactly the (ieta and zeta) parameters? I still don't have the clue of these two parametrers.

ANSWER: Look at the figure and you will realize that these are coordinate axis (like x and y) in the aperture plane.

QUESTION

1. when focusing, we used the fraunhaufer diffraction pattern t calculate the field, this is due to the drop of the quadratic phase term, ok. but this is right only if we make focusing in x and y, what if we just make the focusing in one direction only, as in this case the quadratic phase term is still found in the non focused direction?

2. I can't still understand the validation conditions for usage of fraunhaufer approximation. I mean we for example take ieta = the width of all the rectangulart elements on the aperture like example one in the sheet or the whole width - kerfs or other ???

3. You said in the lecture that, max number of the wires of the probe must be less than 256-512, how comes while in 1.75D you may have 6N or 5N channels which is >>> from 512 if N = 128, and what about the 2D in this case?

4. When calculating the field at some point, are we concerned with just the distribution of the field U(x,y) or the Intenesity of the field I(x,y) = |square(U)|. and why this function is only in terms of x and y althought z is introduced after replacing fx and fx with (x,y/lamda.z)?

5. Regarding the Parameter estimation in CFI, i didn't try to think about the derivations, all i got is that we need to make a parameter estimation of the pixel value because we can't make spectrogram for whole pixels in the image, and we can make this parameter estimation in the freq. domain while the time domain approaches has several advantages relkated to the computation complexity and also time domain is robust in low SNRs so we go throught it and then to improve the estimated parameters we use the auto correlation and cross corelation where some advantage here can't be there. I don't know if this considered as deep understanding or not?

ANSWER

1. You always have focusing in both x and y.

2. whole aperture size.

3. People do encoding methods inside the probe to allow for that number of wires from the probe to the system regardless of the actual number of channels.

4. for a given field, z is a number not a variable. You care about the field U(x,y) and I(x,y) is dependent on it and cannot be computed directly.

5. This is not deep but should be enough for the exam.

QUESTION

For the **coded excitation scheme**, we use long pulses, so signals received may overlap & the axial resolution is lost, but due to using encoding / decoding scheme , we can resolve these signals & separate them again by correlation [template matching].. IS THIS RIGHT??

ANSWER: YES

'Rate of generation of **2nd harmonic** is proportional to p square'... WHAT IS p??

ANSWER: pressure

For the **focusing theory**, I see that the quadratic phase multiplied by the aperture is only cancelled by the lens when the required depth equals the focus... So how we can use the far field approximation for any depth when a focusing lens is applied?? [is it that due to using dynamic focusing, we can focus at any depth so we can use that approximation in this way]

ANSWER: This has a theoretical derivation in the same textbook you has earlier chapters from and is correct whether or not you are at the focal depth.

DOPPLER

In PW Doppler, do we get **sample depth** by the PRF & the **sample volume** by the pulse length?? & if so, why [what is the correspondence]??

ANSWER: Sample depth determines PRF but Sample volume has nothing to do with pulse length in theory.

Why the Doppler **spectrogram** can't be computed from a single excitation [then how the CW Doppler works]??

ANSWER: CW Doppler cannot be called "single excitation". In PW-Doppler, you should refer to the theoretical derivation to realize that the bandwidth of a single excitation is 100 time more than the expected Doppler shift.

Is the **difference between CW & PW** that PW enables localization but may suffer aliasing [i.e. in both, we compute the spectrogram using the same signal processing techniques]??

ANSWER: Yes.

Why when we increase the sample depth we need to decrease the PRF??

ANSWER: Sample depth determines the time between excitations or PRP. PRF=1/PRP.

Is the PW Doppler probe a single element or array [I see that it can't steer or focus if it's a single crystal]?

ANSWER: PW-Doppler uses the same probe as B-mode ultrasound.

I found the materials about Doppler that 'in CW → the Doppler shifted frequency is compared to the transmitted frequency, while in PW → each received echo is compared to a similar echo resulting from the previous transmission'… WHAT IS MEANT BY THAT??

ANSWER: We are estimating the Doppler shift in both cases using different methods. What you are saying is confusing and might not be accurate if you are not using a correlation based method for PW-Doppler. The main difference between the two is that one has sampling of the acquired continuous domain version of the Doppler shift while the other acquires samples directly with a much lower rate (and hence, may encounter aliasing).

For the slide about '**data acquisition in CFI**' that has 3 figs [mechanical scanning, electronic packet scanning & electronic continuous scanning]:

1. Does each point in the figs represent a complete scan line in its direction, not only a pixel?? Yes
2. Why mechanical & electronic continuous scanning need no settling time for the clutter filter?? Incorrect question – completely unrelated!!
3. Is there a good material to understand this part?? This is a practical part that will likely have no material available

ANSWER (repeated from above):

All these methods attempt to strike a delicate balance between getting the max PRF while avoiding any blocking in the image due to having information for different parts acquired at long different times. So, there is no one method that can be considered optimal and each method has its pros and cons.

Is it right that **clutter filtration** for Doppler → before ADC, CFI → after ADC [as high end ADCs with large no. of bits that can cover the dynamic range are allowed for the CFI equipment]??

ANSWER: Yes in general.

For **CFI estimators**, is it right that

4. Autocorrelation → between the RF signal received from a pixel & itself
5. Cross-correlation →between the RF signal received from a pixel due to a transmission pulse & the RF signal received from the same pixel due the next transmission pulse?? [I say a pixel as it's the gate for CFI → 2D multi-range gated]

ANSWER: Yes

Do we consider the **axial resolution only in CFI** [for signals coming from pixels on the same line not to overlap], while we consider the **time & frequency resolution in CW & PW** [due to using windowing for applying STFT]??

ANSWER: Not correct. CFI only computes the mean velocity but has to care about both resolutions.

Can we make a good estimation for the **sonation angle** when using visual aid in Doppler like steerable CW & PW [& thus can recover the 2D velocity vector, not only the axial component]??

ANSWER: You do not estimate the angle. The user input what he/she sees as the angle and then it is used to compute the in-plane velocity.

What's meant by **radial averaging** in auto & cross-correlation for CFI??

ANSWER: Not given in this class.

For **phase aberration correction**, do we use focusing & steering delays in transmission, & use cross correlation between RF signals received at crystals to recover the effects that we applied at transmission [i.e. the focusing & steering]??

ANSWER: No. You do not do anything in the transmission part.

In **STA with spatial encoding**, do we store, for each receiving element, the 4 received signals & arrange them in a vector & then apply the inverse of the decoding matrix to that vector to resolve them in another vector containing the contribution of each transmitting element [so that I can apply beam forming] ??

ANSWER: You receive the weighted sum of 4 beams that you can separate later with the inverse transform. There is no further beamforming after decoding.

What's meant by 'beam forming delay value for focusing is quadratic across the aperture'??

ANSWER: Look at the focusing delay equation and you will find that it is quadratic with element distance from the reference element.

Is it meant by 'array probes can't send & receive at the same time' that the aperture is used in only one mode; sending or receiving [that was a T/F question in the sample exam]??

ANSWER: Yes.

How '**concurrent multiline acquisition'** is done [i.e. how many elements are used for sending & receiving, is sending defocused, do we use single elements or apertures in sending]??

ANSWER: You are using nonoverlapping apertures to acquire line simultaneously and you add them together.

For **Q.1 in the sheet**:

6. Can I assume that I use a focusing technique that enables me to focus at any of the 3 given depths [like dynamic focusing], so that I can use the far field approximation without checking for the depth condition??

   ANSWER: in a problem like this, you should follow the instructions unless there is a missing parameter. You cannot assume dynamic focusing unless you are told to.

7. If that isn't allowed, I found 3 conditions for the far field approximation:

1. The condition written in the slides [that involves eta & zeta]
2. A condition called 'antenna designer's formula' → $z > 2D^2/\text{lambda}$
3. The near to far field transition $L = D^2/4.\text{lambda}$

ANSWER: There was a similar question about this above that you can check. To summarize, it is up to the designer to choose while making sure that the conditions are met.

Which one of them can I use [they give different values]??

8. For (a) → is zeta the whole aperture width & eta the element height [which isn't given; can I assume it 1 mm]?
9. For (b) → is zeta the whole aperture width & zeta the circle diameter??
10. For (c) → are both zeta & eta the biggest diameter?

ANSWER: Should work this way.
