Acoustics & Noise Control
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Module – 18
Lecture – 23
Microphone

In the last class we introduce basic instrumentation workers for a acoustic measurements that is the condenser microphone. We understand there are many different kinds of microphone the condenser microphone is the one that is most suited for the application of acoustic measurements.

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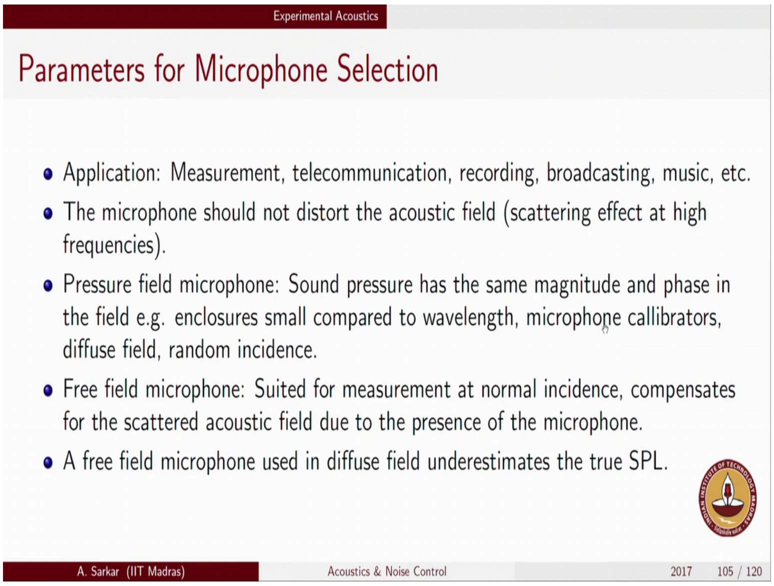

So, carrying on we also sort of touched upon the idea that there are 2 basic types of microphone the pressure field microphone and the free field microphone the pressure field a sometimes which is a slight variant of a diffuse field microphone also that is used in the case when sound pressure is required to be measured in a small enclosure and diffuse field microphone as the name suggest would be more appropriate in a larger enclosure.

But free full microphones are the ones typically which will be used in any quake chambers and there in it is required that is specially designed and for the application wherein it faces only one incident wave and therefore, it is most suited for the applications where it is measurement are done in an anechoic chamber with sort of simulates this anechoic environment.

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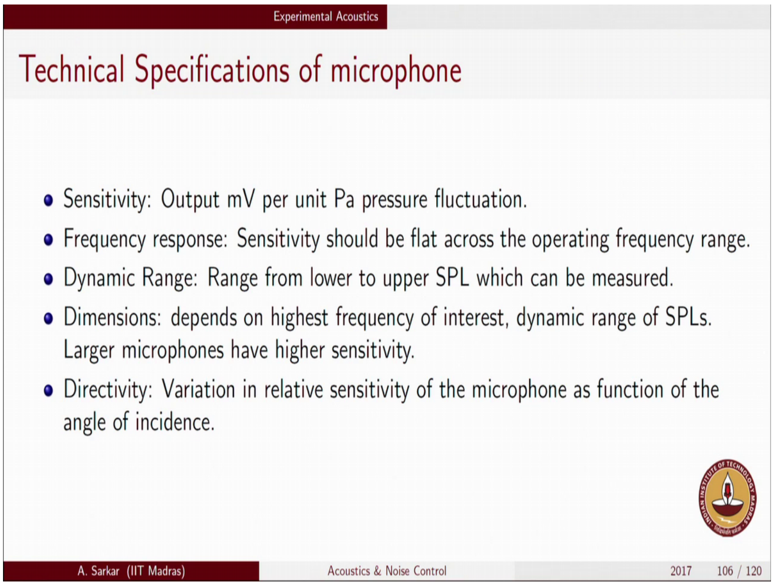

So, moving ahead let us look at the technical specifications of a microphone the microphone just like any other transducers essentially converts any the physical quantity into a voltage and that is the voltage which is measured in electrical terms. So, sensitivity is this conversion as remember acoustic pressure is primarily it is a sort of pressure which is fluctuating. So, fundamentally speaking therefore, for a unit Pascal that that is being for a amplitude of the acoustic wave being a unit Pascal the question is how many millivolts will be generated by the sensor and that ratio the output by input ratio the input to the sensor is the acoustic pressure amplitude wise and the output of the sensor is going to be the millivolt which is generated per unit pressure that is incident on to the microphone surface.

So, this ratio will be called as sensitivity and typically this is one of the very important factor in deciding or selecting a microphone because on one hand you will need a very high value of sensitivity because you do not want these value of sensitivity to be typically very low if it is very low if the voltage is very low then it will be embedded in the electrical noise of your measuring system and it will not be properly manifested right. So, a higher values of voltage typically will give you what is called at known as good signal quality and therefore, a higher sensitivity is what is called for an application and also what is required is that the sensitivity should be flat across the frequency spectrum will come to that aspect in the next this thing the frequency response is the sensitivity factor which is the output by input of your sensor this sensitivity factor if you try to understand if the pressure is at a particular frequency then the well voltage also will be at the same frequency because essentially the sensor works on the principle of linearity.

You have to assume that the I mean it is for it the way the sensor is design is it is supposed to work for linear transfer function between the voltage output and the pressure input right. So, the point is therefore, that if the pressure input can happen across various frequencies which is typically the case in acoustics you are not interested to measure just one frequency, but you are measured you are interested to measure all the frequencies that are there in the sound source that you are interested in. So, therefore, we would like this output voltage also to manifest the same characteristics in terms of voltage as it was in terms of the pressure.

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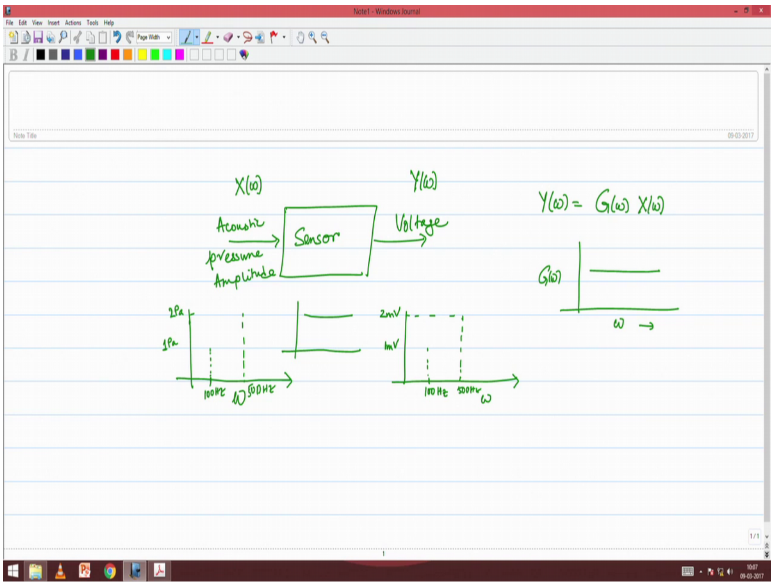

So, sensitivity I will just illustrate myself in notes here. So, a sensor is basically a transfer function the input to the transfer function is the acoustic pressure amplitude and the output is the voltage right, but then these acoustic pressure amplitude can have can be a function of frequency omega right. So, the voltage also depending upon at which frequency the acoustic pressure amplitude is incident on to the microphone sensor the voltage also will pick up exactly the same frequency. So, if the frequency of excitation is this then the frequency of excitation of the output of the sensor would be exactly this because a sensor is supposed to be linear the transfer function relating the voltage to the acoustic input is essential in linear. So, therefore, the frequency cannot change.

But now lets us say a unit one Pascal I mean one Pascal I know is unrealistic in acoustic terms, but just for the sake of understanding the sensor its understand let us take this hypothetical example suppose one Pascal at this frequency let us call this as 100 hertz generates 1 millivolt at the same frequency of 100 hertz right that is what it is, but now let us see in the excitation you have 2 frequencies 100 hertz and the other let us say 500 hertz. So, 500 hertz possibly has another excitation which is double of it. So, it has 2 Pascals it is imperative that the censor reports the voltage for the 500 hertz excitation to be exactly 2 millivolts.

This is if this does not happen if the scaling of the input is not preserved in the output then virtually this sensor is useless in terms of measurement of multiple frequencies it will be useful only for measurement of signals which are having a single frequency and that to you cannot compare between the 2 frequencies that is that you would like to measure or something. So, virtually the sensor will be useless unless it preserves this scaling right. So, the only way in which this scaling would be preserved is to have a frequency response function which is flat. So, the transfer function relating let us say this is my input X in the frequency domain the output is denoted as Y in frequency domain. So, the transfer function will be Y omega is equals to z omega times X of omega.

Similar to what you would have done even in Laplace transform case, but since we are taking steady state the question of Laplace transform does not arise frequency transform is good enough right what we want is that this g omega function across omega should be flat if it is anything, but flat then what I just illustrated in terms of scaling of the between the input and output will not be preserved in which case certain frequencies will unnecessarily be undervalued or overvalued in terms of voltage though in terms of pressure they may have let us say you are talking about a white noise situation. So, in terms of pressure all frequencies have equal contribution, but if this g omega associated or the transfer function associated with the transducer is not flat then you will not get a flat spectrum in the output which is the voltage corresponding to the transducer even if the input is flat right.

Which means you are going to make an incorrect measurement we which means that some frequencies will be unnecessarily underestimated or overestimated depending upon how the transfer function behaves right; so, this is something which is very important and this is what we will try to look into in great details. So, the frequent one very important attribute your qualification of a sensor which you must never compromise on is that at least for the frequency range of your interest you should make sure that the sensitivity chart corresponding to the microphone that you are using is showing a flat frequency response and remember you are not interested to have a flat frequency response from 0 hertz to infinity hertz.

Anyway as you know we hear technically it is said 20 hertz to 220 kilohertz, but possibly the range of interest in your industrial application could be a subset of that possibly after 50 hertz only you will be interested; so, 50 hertz to maybe 15 kilohertz. So, if the frequency response graph shows to be reasonably flat in this region you should be happy with it if you are very sure that you are looking at an application which does not generate of the frequency is in the higher range or in the lower range then appropriately you can go for another microphone also which compromises on those ranges which are not in the criteria of your measurement, but you should not compromise on the frequency in the sensitivity within the frequency response that you are interested.

That is the bottom line right and we will see how what are the restrictions what are the implications of the size of the microphone and the design of the microphone in terms of this sensitivity aspect and then another important attribute of the sensor is that of a dynamic range. So, you will typically be interested to measure different range of sound pressure levels using the same microphone you should I mean you would possibly demand from the microphone supplier that it should be able to measure very feeble sound to a very loud sound, but sometimes what happens is that if there is a loud sound associated with that there is a large voltage and use and the sensor manufacturer usually will have some cut off limits.

Because you will not allow a large voltage in his circuitry because that will get spoil. So, again depending upon what is the range in which you would have an estimate depending upon your application that what is the range of s p l values that you typically would like to measure. So, you should choose microphone which allows for that range of measurement you should not use a microphone which is meant for you know measurements in a recording studio of this sort in an application such as aerospace where you know that there is an intense sound that is generated. So, microphones meant for aerospace applications like aero engine noise or things like that will be made with special care such that they can handle that very high intense level of noise.

So, dynamic range is another crucial factor as I said dimensions are very important the if typically you will if you recall the last lecture that we said that if you have a larger microphone then you can expect due to the presence of the microphone the acoustic field will be distorted and therefore, the measurement will be corrupted you would like to measure the acoustic field as it is sort of in the absence of the microphone, but they just mere presence of the microphone as a solid object will create a certain boundary condition around at least the region of the microphone and therefore, microphone will be able to measure not the incident field, but rather the distorted field caused as a result of the presence of this microphone.

So, therefore, it is imperative to make sure that the level of distortion is exceedingly small or sort of can be ignored in a real application situation and that happens when the microphone dimensions are much smaller compared to the wavelength of the sound that you are measuring. So, again from the calculation of the frequency range of interest you should be able to find out what is the wavelength of the sound that is expected to be measured by this microphone and then you should check that the dimensions are far smaller dimensions of the microphone are far smaller than the dimensions of the wavelength of sound and the other way in which it effects is that. So, in some sense smaller is better in terms of the scattering effect, but larger is better in terms of the sensitivity effect.

So, that is interesting that is what these are the 2 sort of competing factors which influence the design of the microphone itself and as I said at least for free field microphones they are designed to work based or they are sort of calibrated to work under the situation where the acoustic pressure is normally incident on to the surface of the microphone the diaphragm as we will call it. So, it so happens and as I illustrated schematically at least in the previous talk if the direction of the incident wave is not normal then it can cause changes in the in the response in the mechanical response of this diaphragm structure on the microphone and as a result it can lead to a difference in the sensitivity of the microphone. So, directivity of such prefilled microphones is going to be very important and typically the manufacturer will also report this it is calibration in his literature that what is the variation of the sensitivity if this microphone is used for various for measurement which is such that the incident acoustic wave is not completely at ninety degree, but at a direction.

But then as I said that it is recommended that you use the microphone in the incident direction itself in the direction normal to the incident wave, but sometimes the problem is you may not have a very accurate estimate of this direction. So, you would like to know that what are the range of possible errors that you will make in case you go off. So, you typically like to have this sensitivity change within certain bounce. So, that is what will get reported, but please understand that this issue is there with free field microphone that it has a directivity and it is recommended that it will work best only if it is facing the acoustic wave in a normal fashion this issue is not there in pressure field microphone because anyway we understand that the pressure field microphone is supposed to work in a sort of omni directional fashion we are saying that if you are interested to measure sound in a small enclosure where like let us say within a resonator in such a case we are expecting that anyway across the entire points of interest the pressure field does not changed by much in which case it really does not matter.

So, let us now look at how.

Student: (Refer Time: 16:20).

Yeah is called sensitivity gain (Refer Time: 16:24) control system terminology yeah.

Student: (Refer Time: 16:28) still be able to interpret what is the (Refer Time: 16:30).

You will be inter we will to interpret only if it is a single frequency, if it is a multi frequency combination you will lose the interpretation right.

Student: Even then the frequency of the output signal it is (Refer Time: 16:44).

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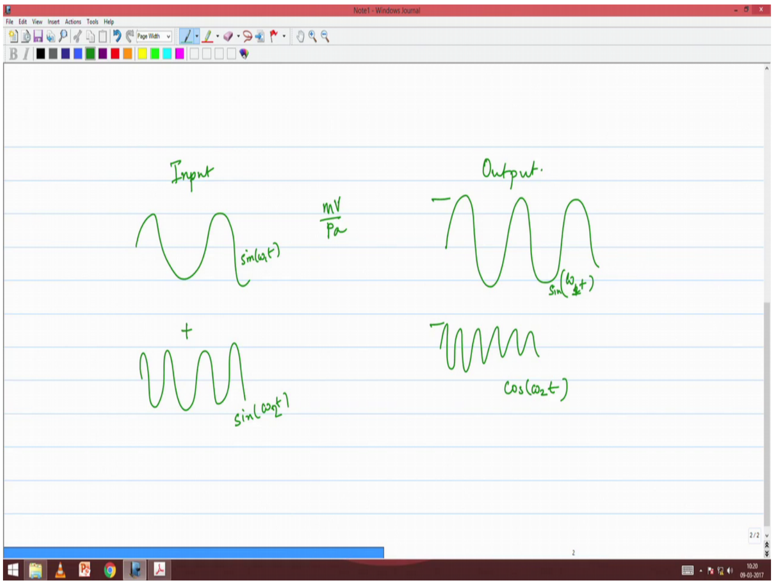

Yeah, yeah, if it is a single frequency input then you will get a single frequency output I think I understood your question. So, the point is this that even if see even if this sensitivity is not flat what will happen is that when you have only this I think I will re do in the next page if you have a single frequency input you will always have a single frequency output right. So, this is irrespective of the sensitivity right, but if you have a combination of 2 frequencies let us say these 2 frequencies are of equal amplitude you would expect the corresponding voltages also to be of equal amplitude only then they will combine in the right fashion right.

If it. So, happens that despite these 2 frequency input frequency is being at the same amplitude the output here is sort of undermined right then in when you look at the voltage signal in your oscilloscope you will see that the high frequency is not. So, important low frequency is more important so; that means, what will sort of contradict your actual this is the input is actual right the reality the output is what you will see right typically you are interested in a source where there are multiple frequencies right otherwise if it is just a single frequency measurement it really does not matter as you say, but you will not be interested to buy a microphone to measure only 500 hertz sound all right or 1 kilohertz sound.

Student: You can still (Refer Time: 18:32) you know what is the gain function then we again we can to the voltage output that we get.

No, but see there is a the way it works is that you put a calibration factor right this whatever is this millivolt per Pascal this value that is a single value that calibration value is a single value. So, therefore, each of them will be interpreted I mean you just convert. So, whatever is the millivolt value here you just convert this back to the Pascal value using a single value that is the way it works and the other issue that I did not mentioned in this talk is that a phase also see if the phase here is going to change then also it may cause and add let additional subset let us say this is sin omega t sin omega 1 t this is sin omega 2 t right sorry sin omega 1 t. So, the phases not changed and this one was sin omega 1 t and this is now become cos omega 2 t sorry this is 2 right.

So, this will now be a sin and cos combination right the sin plus cos combination is not going to give you a sin plus sin combination the amplitudes again will change the r m s also will change right. So, the phase distortion also between the 2 frequency should not be there that is called phase distortion.

Student: So, 94 also distorts the phase (Refer Time: 20:02).

Any general transducer can be treated as a gain function as you said. So, that gain function should be flat without any phase distortion in that case the calibration is very easy right see another question that will come is that of calibration which will touch upon briefly when we calibrate a microphone we calibrate for not for all frequencies we can calibrate mostly will calibrate for just one frequency right we cannot calibrate once for omega 1 once for omega 2 and so on and so forth what you are saying that instead of calibrating for a single number you calibrate for the graph. But that requires your microphone for every measurement you need to recalibrate the sensitivity chart which is reported is just like a benchmark which is there at is nation state right, but usually in operation you will have a calibrator which will calibrate this sensitivity value at the start of the experiment and this is going to be a single value.

So, unless you are sure that the frequency response is flat this no point in doing that calibration or you have to do calibration across at least all the octave bands which will go be a very tedious process all right. But yes in principle you can think that the calibration can be done in terms of a graph also, but that will make things pretty much complicated in actual working out in actual execution of the experiment you will require calibration across all frequencies now let us look at the condenser of microphone.

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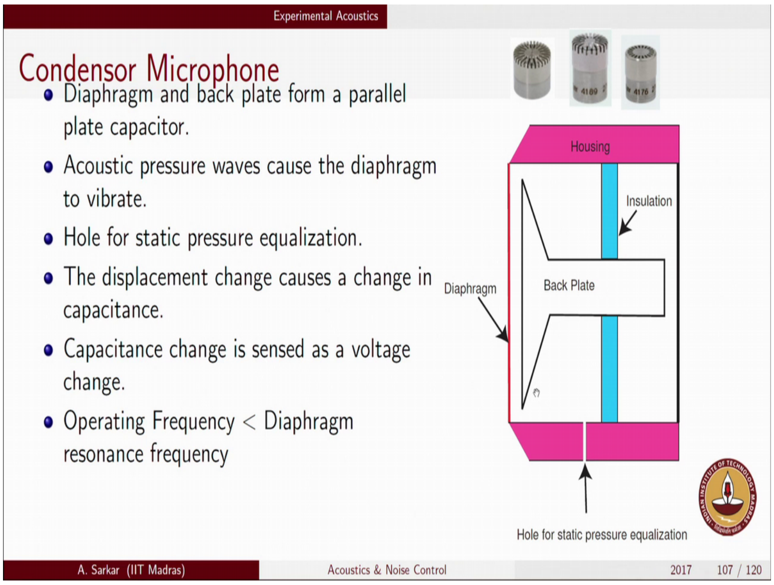

what you see in the top are pictures taken from the blue whale and care website these are how condenser microphone looks like possibly these are not of the same thing that you would have looked when you would have used a microphone for singing a song or what you have regularly see the microphone that you are seeing in my caller right. Now it is not of this kind it is a condenser microphone exclusively design for measurement applications will come to the those performance microphones which are used by singers in a little while.

But this is how it looks at least from the outside and they come in all different sizes I mean it is like 1 by 8 inch 1 by 4 inch these are the typical sizes of these microphones now internally this is how it looks like and this is just a schematic diagram which I have drawn myself. So, this is how schematically the microphone looks from the inside this surface here is basically the diaphragm right what and this is the surface which phases the acoustic waves. So, the acoustic waves are incident on this diaphragm surface and there is a housing which is pretty rigid which sort of on which this diaphragm is mounted and then there is a back plate here.

So, this is all this part is also rigid. So, what happens is that between the diaphragm and this back plate there is a capacitor that is at work right and you know the capacitor basically works on the principle that it has to be charged because there I mean the charge capacitance and the voltage across the 2 plates are related by the formula which is of the kind q equals to C v right. So, in that case what we have is that the capacitance in itself depends upon the distance between the diaphragm and the back plate rights. So, there is the capaci parallel plate capacitor which is formed between the diaphragm and the back plate and the capacitance itself depends upon the distance of separation between the diaphragm and the back plate.

What happens when the acoustic pressure is incident on this diaphragm is that the diaphragm will start vibrating and when it starts vibrating the distance of separation between the back plate and the diaphragm will change. So, if the distance changes the capacitance changes and if the capacitance changes then and provided the charges kept fix the voltage measured across this 2 phases will change right. And if you can therefore, pick up the voltage change you can calibrate this voltage change to the change in separation distance between these 2 parallel plates and that intern can be calibrated to rid off what is the acoustic pressure which has caused this change in distance.

So, that is basically the principle in which this transducer will work will do the detail calculations shortly also, but let us understand a few more important things the first thing is that this diaphragm should not have a static deflection in the static condition it should have a certain profile which; obviously, should be like plain and simple plain air profile right you can expect that the deflected profile will not remain plain and it will deflect somewhat like a sinusoid. So, it is you could not possibly construct this diaphragm within a vacuum filled cavity had it being vacuum cavity on one side then it would have deflected under atmospheric pressure right. So, therefore, you need a atmospheric pressure to be existent on both the sides of the diaphragm such that the static deflection is 0 of this dia of this mechanical structure called the diaphragm.

So, to ensure that the static pressure is 0 we should not have this cavity in vacuum rather it this cavities should have atmospheric pressure. So, towards that in there is a hole which is there an; obviously, this is just a schematic description of the hole this is not actually how the physical hole is there right. So, depending upon the manufacturer there will be variants, but at least schematically we should understand that there is means by which atmospheric pressure is maintained on both sides of the diaphragm on the outer side nothing needs to be done it is all these atmospheric pressure on the interior side you must have this atmospheric pressure created and or the static pressure equalization done and that is done with some sort of a hole.

As I said the displacement here bit of the diaphragm once it is excited once any acoustic pressure wave is incident on to this diaphragm structure the acoustic pressure wave is a dynamic loading it is like a pressure loading which acts on this mechanical structure which is diaphragm and you can easily calculate at the how much is the displacement of this diaphragm due to that pressure loading either using simplistic vibration theories of lump model or if you think you can infant employee finite element method type simulations. But usually finite element methods are not required because remember we are talking about the situation where the structure is much small compared with the wavelength of the sound.

So, basically since the structure is small the lumped parameter approximation theory works pretty much well for this sort of a structure and that is what we are going to take up in the next few slides. So, we are actually going to find the displacement change due to the pressure loading that is actually incident on to the structure and then we are going to say how this displacement change causes a change in capacitance and then finally, how this capacitance change causes a change in voltage right. So, that is how things will work. So, capacitance change will be sensed as voltage change and as we will see we will actually find out the sensitivity associated with this condenser microphone we will see that the sensitivity will remain flat only below the first natural frequency of this structure below the fundamental frequency of the structure right.

Once you are approaching the frequency which is near about I would say like half of the diaphragm natural frequency then you are surely going to see that the sensitivity will get distorted it will no longer be flat it will pick up and then even if you cross the resonance you will no longer get a flat response and that will what sort of distort the sensitivity.

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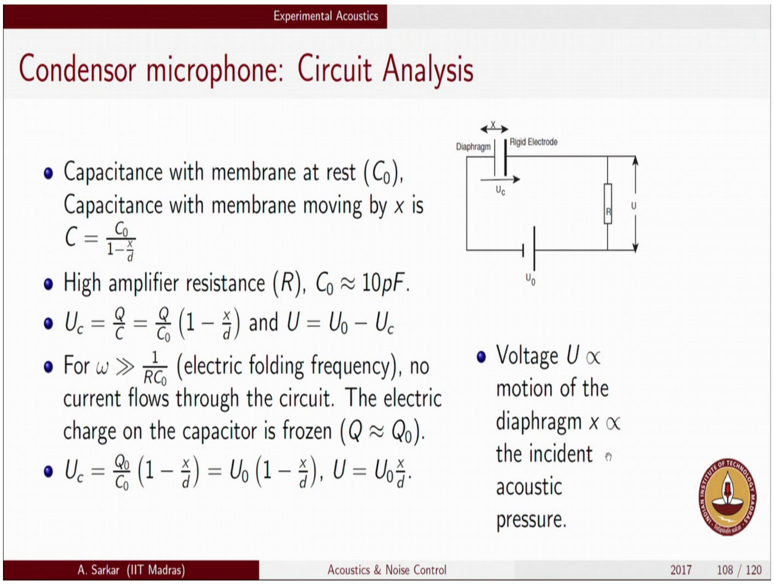

So, let us look at the electrical circuit now as I said to mention to measure this capacitance change it has to be polarized you have you must have a dc source which puts up an initial voltage towards this parallel plate capacitor because if you do not have any initial voltage there is no charge. So, does not work like a capacitor if there is no voltage this no current or charge that will be coming out.

So, to make this work you need to have some source of polarization sometimes it is an external polarized source sometimes it could be like just a battery kind of things some sometimes like even in collar miles the way sometimes it can happen is that these, these are called electric type microphones wherein right at the time of manufacturing they ensure that this charge between the 2 plates are residing right and that is what happens in other microphone classes, but let us stick with condenser microphones used for measurement. So, usually most of them would be requiring a polarization voltage. So, this is the polarization voltage. So, polarization voltage could be again either a dc source or you can use your ac power supply and then rectify it and use an adaptor which will basically ensure that initially before the wave is incident I mean the starting condition of this transducer is that there is a residing voltage across the 2 plates of the capacitor that is a diaphragm and the back plate which essentially ensures that there is a charge equal and opposite charge residing on the 2 plates.

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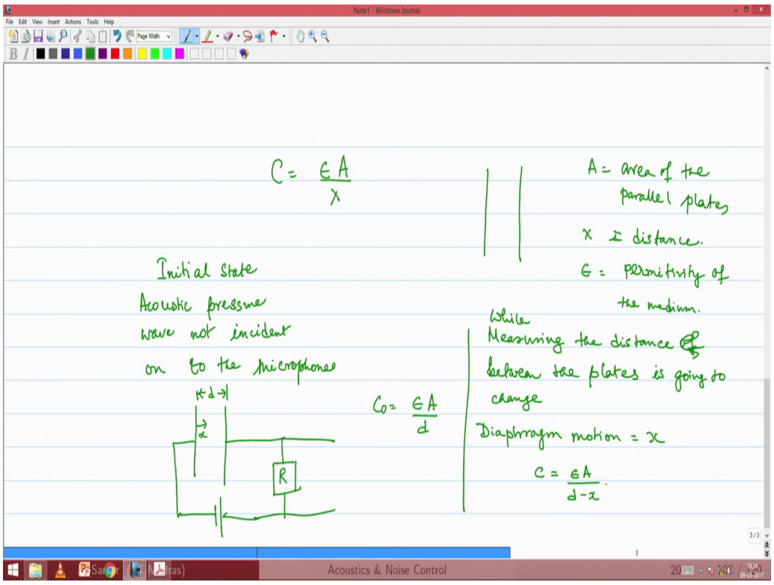

Without that the capacit these condenser microphones will not work by principle and if you recall the formula for the parallel plate capacitance is going to be something like this epsilon s by X right or a I should call this a is the area. So, the capacitance between 2 parallel plate capacitor is given by this formula a is the area of the parallel plates and X is the distance right and C is the capacitance epsilon is the permittivity of the medium since it is air it, it can be found out very easily what is the value of this permittivity right. So, let us understand 2 situations one is the initial state in the initial state the acoustic pressure wave is not incident pressure wave not incident onto the microphone. In other words the acoustic source is not yet switched on right just you have polarized the system which means that initially between the diaphragm and the back plate you are having a certain voltage right you are having a certain voltage and because of the electrical circuitry there is the way it is design. In fact, there is a resistance also and these are the open terminal switch are coming out from the transducer.

So, basically voltage will be measured across this terminal. So, that you can change sense the change of capacitance and thereby sense the change in the distance between the 2 parallel plates. So, the distance between the 2 parallel plates is initially denoted by d in my notation. So, the capacitance is epsilon a by d in the initial state and in the measure while measuring this distance is going to change because a diaphragm is going to move in response to the applied acoustic pressure. So, while measuring the distance of sorry not of between the plates is going to change by how much by exactly the motion of the diaphragm back plate is supposed to be static back plate is not moving back plate is static. So, we are going to assume that the diaphragm motion is X.

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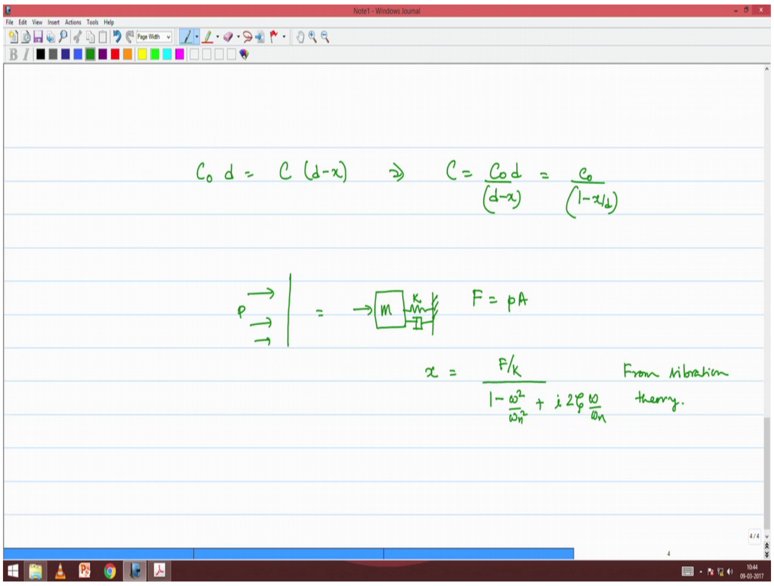

So, this diaphragm is going to move by X which means capacitance when the acoustic source is switched on is going to be epsilon a by d minus X right. So, at the end of the day what do we have combining these 2 conditions the combining these 2 conditions we have C 0 into d is equals to C into d minus X which means the capacitance when the microphone is actually measuring the incident sound is equal to the initial capacitance when the acoustic source was switched off divided by d minus X into d right. So, that is precisely the formula that I have quoted.

C is equals to C 0 1 minus X d. So, that part also can be done pretty easily. So, this seen one more step could be written as C 0 is equals to 1 minus X by d I am just divided the numerator and denominator by d right now note by design what we will have is that X will be much lesser than d. So, the initial separation that we will choose to keep is going to be much more than the expected order of movement of this diaphragm there are quite a few reasons for that one reason; obviously, is that you do not want this diaphragm to actually make a mechanical impact with the back plate right in that case the diaphragm will definitely suffered some catastrophic failure. And even otherwise if you make it very close the electrical field will get distorted and you know because actually strictly speaking the electrostatic force will be very high if these 2 parallel plate capacitors ten to come very close together right.

So, those complications will be there. So, at this stage it suffices for us to understand that the capacitance of this in the measure in the state in which it is measuring is definitely given by this formula and X being the motion of this diaphragm is going to be much lesser than the initial separation between the diaphragm and the back plate which is denoted by d. And as I said this resistance in this circuit is going to be chosen very high where as the initial capacitance C 0 is usually of the order of few picofarads now what we have is that u C which is the voltage across the 2 parallel plate capacitors is going to be q by C the amount of charge which is residing in the 2 parallel plates divided by the capacitance of the parallel plate capacitors and this is C by C v mean the capacitance when it is when the acoustic source is switched on; that means, there is a certain motion.

But C can be related to the initial capacitance wherein the source has not yet been switched on using these formulas. So, if you substitute that back we get C equals to C 0 1 minus X by d right and also applying the voltage loop here the voltage that you see across this terminals is denoted as u this is u C the voltage between the diaphragm and the rigid electrode is u C and the dc source is u 0. So, this u is u C sorry u 0 minus u c. So, u C and u give gives one loop and u 0 gives another loops. So, u 0 in other words this change in potential is compensated by u C plus u u is the potential between these 2 points in the circuit. So, this is just the voltage equation in the circuit.

Now, comes an important aspect that if the frequency is 1 by r C 0 is greater than the electric folding frequency 1 by r C 0 is called the folding frequency of the electrical circuit and we will derive this also in little more details, but the point is this you can take this as an assumption. Now, but will as I said I will probably make this even more clear in the next talk that if we have this condition omega is greater than 1 by r C 0 what is going to happen is that this part of there is actually the effect of the resistance as I said resistance is very high so; that means, if the resistance is high here the charges sitting down here will find no incentive to go around this path.

So, no current will actually flow in this parallel path it will wait here. So, basically there is no flow of charge the parallel plate capacitor itself just has accumulated certain amount of charge, but the charge is not in favorable to flow because of the high impedance associated with this high value of the resistance in this circuit. So, therefore, the electric charge whatever was the charge in the initial state when the acoustic source was actually switched off that charge is remaining constant this is the essential implication of this assumption if you may. So, call it for now, but as I said in the next class I am going to do away with this assumption and actually present you the sensitivity in a more detailed fashion right, but for now you have to take it as this way in a sort of intuitive fashion that the resistance is high if the resistance is high then you must have no charge leaking into this path. So, therefore, there is no other way, but the charge will remain frozen in the circuitry right.

So, whatever is the charge between the 2 capacitors initially that remains constant even when the capacitance changes the charge does not change is going to be our assumption for this simplified analysis will redo this analysis with without this simplification also in the next class. So, what we have as a result is q u C we had already did derived as q by C 0 1 minus X by d with this assumption we now can replace q to be q 0 and q 0 by C 0 is u 0 right because that is the initial condition at the initial state you had just an initial voltage u 0 an initial charge q 0 and a initial capacitance C 0 C 0 by the way is not the same as C.

Because the distance does matter, so, q 0 by C 0 is u 0 and 1 minus X d remains right. So, if u C is u 0 1 minus X d and you also had u equals to u 0 minus u C using these to relation you can say the voltage across these 2 terminals is u 0 X by d in other words if you measure the voltage across this terminals you are essentially measuring the displacement of the diaphragm right. So, that was the key feature. So, if you measure the voltage that can be mapped to the displacement of this diaphragm u u 0 being the initial polarization voltage again that is going to remain constant irrespective of what is the sound that is incident on to this microphone set and similarly d is a constant which is the initial separation between the 2 black plate between the 2 plates the diaphragm and the back plate.

So, therefore, this is the manner in which you can actually calibrate this output voltage to the response of the diaphragm right. So, the voltage u is proportional to the motion of the diaphragm which is denoted by x, but then the motion of the diaphragm is also proportional to the incident acoustic pressure this part is simpler to ask because as mechanical engineers we understand how the motion is related to the incident acoustic pressure the circuitry probably is going to take us a little I would say it is not very comfortable for us to appreciate, but hopefully it is clear which is why I did it slowly. But hopefully it is clear that the voltage is proportional to the motion of the diaphragm the motion of the diaphragm and the incident acoustic pressure the way it is related is no big deal because we understand that the motion of the diaphragm if we look at the diaphragm as just a spring mass system it will have a certain motion it will have a certain force that is incumbent on it.

So, if I just quickly re do this analysis. So, this is the diaphragm and the diaphragm is getting and excitation in the form of pressure the total force that is acting on the diaphragm is p times a essentially the diaphragm is lumped as a spring mass system with mass m and the compliance we are the stiffness being k and; obviously, there is going to be some damping also. So, we already know that the transfer function relating the motion of the system with the incident with the loading is given by this form I 2 zeta omega by omega n this is from vibration theory right I do not want to derive this part of the result, but this is how we are going to we know that the motion of the spring mass system is going to be related to its static deflection the static deflection of the system is just going to be a f by k and the transfer function here which is called the dynamic magnification factor is given in this sort.

So, in the following we just use a notation where we say eta is 2 zeta right. So, with that we are going to get.

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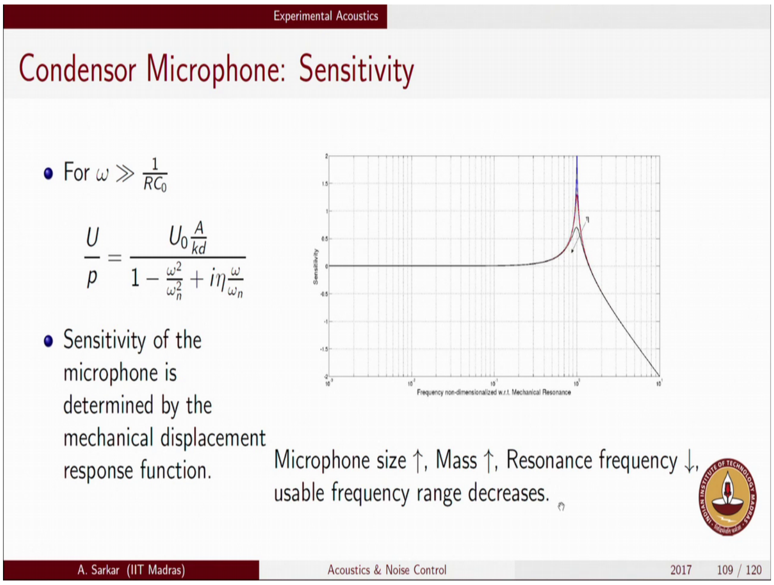

This as our denominator and see p into a is basically going to be our force right the forces p times a. So, u was u 0 X by d and X is p a by k divided by this denominator as we have just said. So, the key factor as I said is sensitivity which is the voltage divided by the incident pressure. So, the voltage divided by incident pressure comes out as this form and what you see here is for a fixed value of u 0 a and k and d all these are design parameters associated with the microphone the only thing that will change the sensitivity is this frequency ratio the ratio of the operating frequency to the mechanical resonance frequency of the structure the first fundamental frequency because this just a lumped system model it will only give the first frequency of the fundamental frequency.

So, the fundamental frequency this ratio of the actual operating frequency to the fundamental frequency dictates the sensitivity together with this eta factor which is basically mapping the damping ratio zeta, but it is exactly not zeta it is twice of zeta right. So, if I plot this graph this is how it looks like as I said it is very very flat starting from these a logarithmic graph I must say that starting from 0 onwards it remains flat it peaks up at ten to the power 0 which is 1. So, at 1; that means, when the frequency operating frequency matches the mechanical resonance frequency of the diaphragm the flatness of the sensitivity graph is lost and also the phase undergoes a sharp change from at resonance the phase undergoes a sharp change from 0 to 180 degree right if you look back at your vibration theory. So, here you show the key point to appreciate is that you should not work with this microphone beyond about this range.

Till about here we see it is reasonably flat we could make use of this microphone, but once this sensitivity start chart shows or behavior where it climbs up and then again it falls down. So, it is not that you know accepting for this frequency we can use it. So, never again it comes back flat right. So, that is the part that we wish to avoid. So, we will by this analysis what is shown is that the operating frequency of the microphone should strictly be below the mechanical resonance frequency of the diaphragm of air condenser microphone right. In the next class we will see how that that electrical folding frequency comes up today we have just taken that as an assumption, but hopefully in the next class we will prove it and establish that there is a lower limit also which dictates the flatness of this sensitivity response.

So, as I said the sensitivity of a microphone is determined by the mechanical displacement response function which is basically this denominator factor what we see here is that if the microphone size goes up the mass will go up if the mass goes up resonance frequency will come down if the resonance frequency will come down then the usable frequency range will decrease right because omega by omega n is what matters you should not cross omega by omega n to the extent of 0.3 or something like that you should not approach near about 1 even 0.5 is looking bad. So, maybe 0.3 which is around here is ok or better still stay at 1 by 10 right.

So, that that is what the guideline we extracted in today's class. So, we will pick it up from here will do a more accurate sensitivity analysis in the next class.

Thank you.