Audio System Engineering
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Lecture - 24
Loudspeaker

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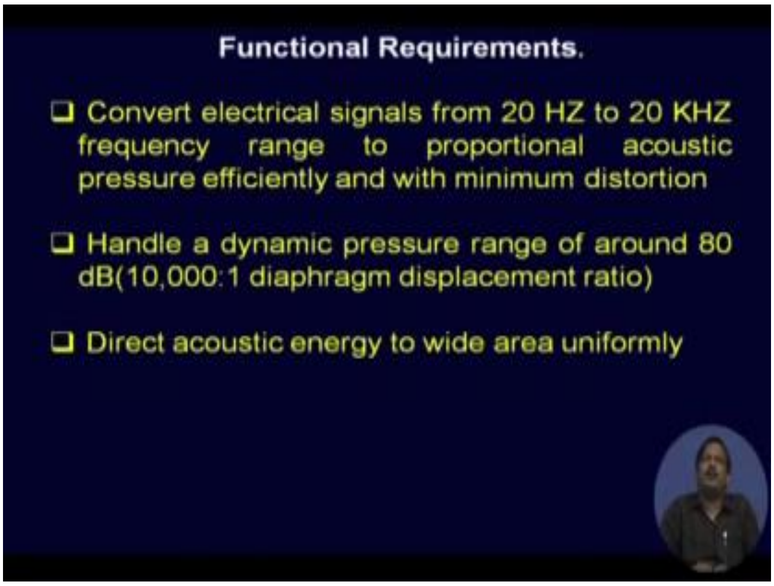

So, now we have start the study about the microphone. Now we go for the loudspeaker design. How do design the loudspeaker. Now, if you again the loudspeaker specification is also important that frequency response and then there is a resistance of the loudspeaker is very important. So, in loudspeaker what is the functional requirement, the requirement of the loudspeaker is that it should convert the electrical signal to acoustical signal. So, convert electrical signal from if you say audio range 20 hertz to 20-kilo hertz frequency range to proportional acoustical pressure efficiently and with minimum distortion. So, if I say the purpose of the loudspeaker is that to convert electrical signal maybe supply from 20 hertz to 20-kilo hertz to the proportional acoustical pressure with minimum distortion, distortion should be very minimum.

So, handle a dynamic pressure range of 80 dB that is that is means 10 to the power 4 is to 1 diaphragm displacement ratio 80 dB. To direct acoustic energy to wide area uniformly, what is the purpose of the suppose I in this room, if I put two loudspeaker the purpose is that what about the acoustic electrical signal is coming from amplifier, it should convert to the acoustic signal and direct in uniformly in the room. So, this is the functional requirement often loudspeaker.

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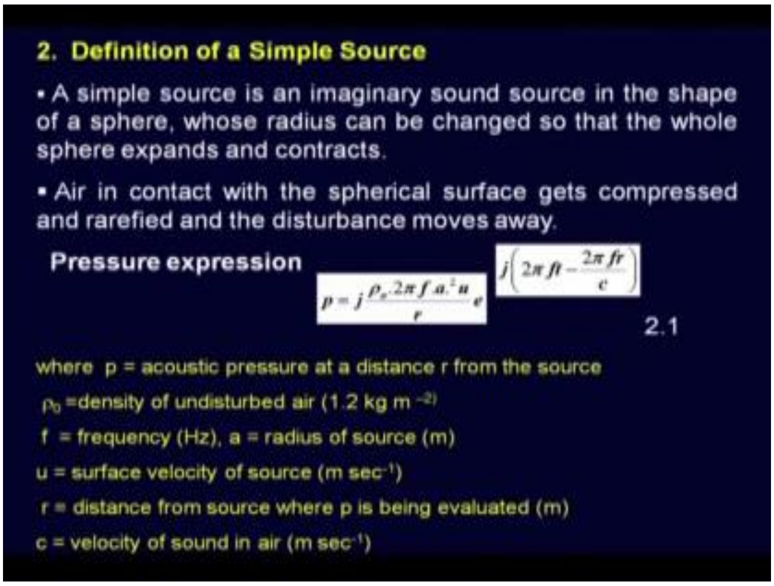

Now, go for the theory. So, loudspeaker is nothing but a source of sound. So, it can be consider a sound source. So, definition of a simple source, what is simple source, simple source is an imaginary sound source theory not in practical, let us theory, a simple sound source is an imaginary sound source in the shape of a sphere whose radius can be changed so that the whole sphere expand and contracts. How the sound will be producing here. So, I can consider imaginary sound source here which is kind of spherical in nature. So, it is change of the sphere whole sphere it change contract and expand and then it produce the sound in spherical way of the sound will be propagated in all direction. So, this is the definition of the source. Air in now if the sphere is changing in contraction and expansion so surround air will also have contraction expansion and that way the sound energy will be distributed in the room.

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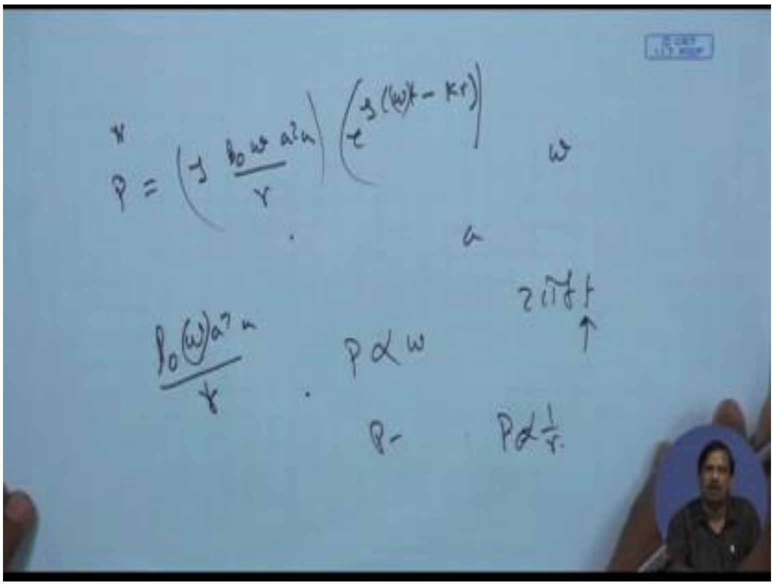

So, expression of the pressure of the sound source will be P is equal to J into rho 0 omega a square u divided by r; r is the distance from the source or e to the power J omega t minus k into r, where P is the acoustic pressure at a distance r from the source. If the p is the acoustic pressure at a distance r from the spherical source then it is nothing but a J rho 0 omega a square u; a is the diaphragm area of the, a is the radius of the sound source is contract and expand. So, a is the radius of the source, omega is the angular frequency. And if you see I have written in term of omega is written term of twice pi f. So, it is defined f is the frequency, u is the velocity - surface velocity of the source, r is the distance from the source where p is being evaluated, c is the velocity of the sound and rho 0 is the density of the and this equilibrium density of the air.

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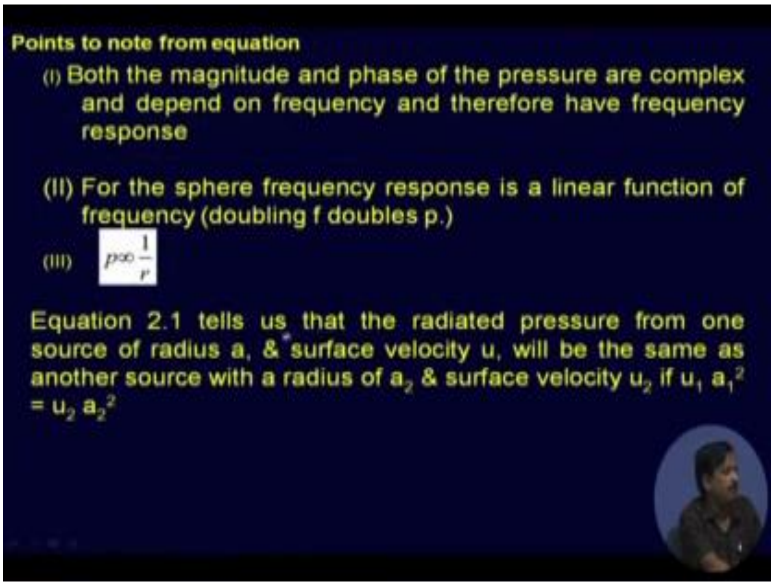

So, if I see in that equation both the magnitude and phase, so this is the phase, this create the phase, and this create the magnitude both are complex and depends on frequency. If I say the amplitude is rho 0 omega a square a, a square u by r that also depends on the omega the frequency and phase is also depends on the omega the frequency. So, for the sphere frequency response is a linear function of frequency that means, if I say the P is nothing but a proportional to omega P is proportional to omega, omega is nothing but a twice pi f. So, if I double the frequency, the pressure also will be double. And P is inversely proportional to 1 by r which is the spherical way propagation because that the inverse square law is support for this only. Now, this equation of the pressure tells us that the radiated pressure from one source of radius a.

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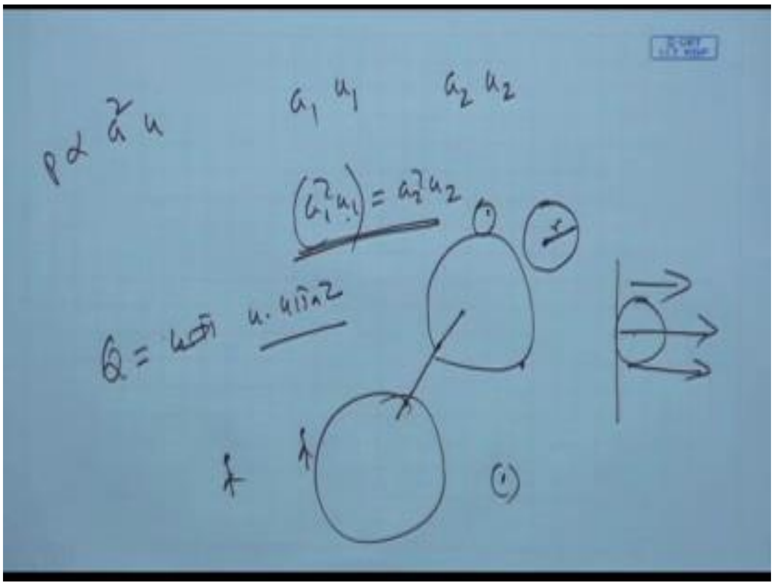

Now if I say and surface velocity u, so if I say radius a, I have a, one source with radius a and surface velocity u. So, pressure is proportional to a square into... So, if I say one source have a radius a 1 and surface velocity u 1, another source has a radius a 2 and surface velocity u 2. Now, if a 1 square u 1 is equal to a 2 square u 2 then the both the source will create the same pressure. So, for one source of radius a surface velocity u then the same another source with a radius a 2. So, with the different radius, if I create the radius of the diaphragm is different someone is very small radius diaphragm, and some large radius diaphragm. If the velocity of the sound is also change, if the surface velocity of the diaphragm you change, if it is a 1 square u 1 is equal to a 2 square u then I can say both the loudspeaker will produce the same sound pressure.

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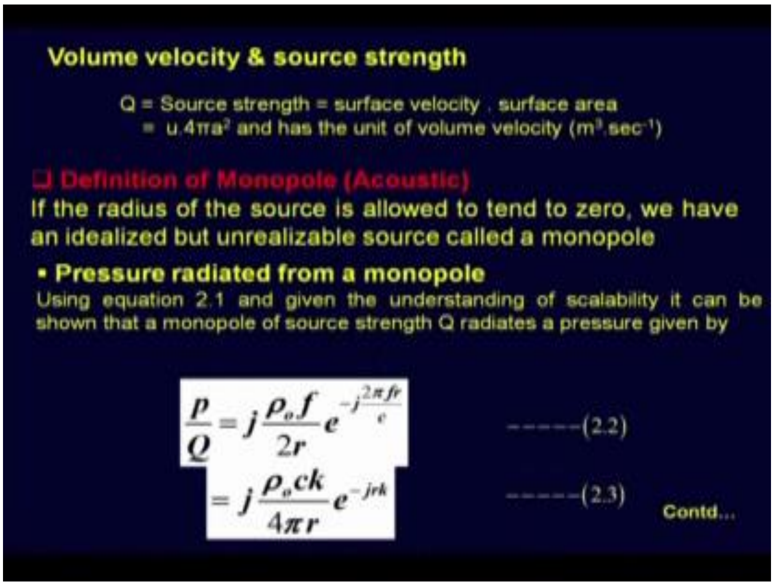

Now, what amount of sound pressure will be produced that depends on a 1 the radius of the diaphragm and a and u. So, now, you call this is call volume velocity or source strength. So, if the other thing is remain constant then the source strength Q is defined as the product of surface velocity into the surface area - surface velocity u into the surface area. So, u into 4 pi a square is called the strength of the source - Q. Strength of the source is nothing but a surface velocity multiply by the surface area; and what is the unit velocity and area. So, it is a meter cube per second, volume velocity meter cube per second.

Now, come to the some definition will go far then then go for the microphone designs what is the definition of monopole acoustic monopole not that micro monopole acoustics monopole. If the radius of the source is slow down to tends to 0, we have an idealized, but not realizable source called a monopole. So, if I want a surface sphere to produce the sound of radius r by changing the radius I create the sound. Now, if the radius is tends to zero point source then I can say it is nothing but a monopole; theoretically it is possible, but it is not realisable. You say the spherical way propagation is fail, solution is fail when an r equal to 0 or r equal to infinity both the way. So, r equal to 0, the solution is fail. So, if the r is equal to 0 then ideally it is not realizable, but ideally it can possible then it is defined as monopole. So, pressure radiated from the monopole can be written as P divided by strength of the microphone strength source strength then it will be j rho 0 f by 2 r into e to the power this thing. If it changes to f is change to c k then it will be c k by 4 pi r square, so it is nothing but a change.

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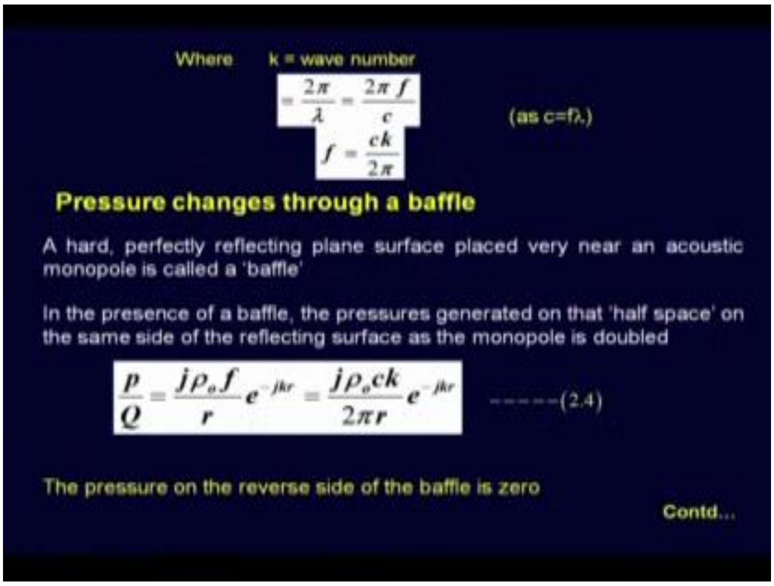

How this can a p is k is the wave number. So, this 1 pi by lambda 1 pi by f by c. So, f is c k by 2 pi. So, if it is if it changes by c k by 2 pi, this will be the pressure divided by surface strength. Now, in pressure change through a baffle, a hard perfectly reflecting plane surface placed very near an acoustic monopole is called a baffle. Suppose, I have a sound source here if I have a baffle then the strength, change the strength of the monopole it change. So, in the presence of a baffle, the pressure generated on that half space on the same side of the reflecting surface as the monopole is doubled.

What is the meaning, suppose I have a sphere is a monopole. Let imaginary thing that the radius of this monopole is 0. Then if I put a baffle here reflecting surface then this side little bit is spherical in nature this side reflection would directed to this sides and the strength of the source will be double. So, if it is monopole strength is then it will be double 2, this is 4 pi, 4 pi r. So, if I multiply by 2, it is 2 pi r c k pi 2 pi r.

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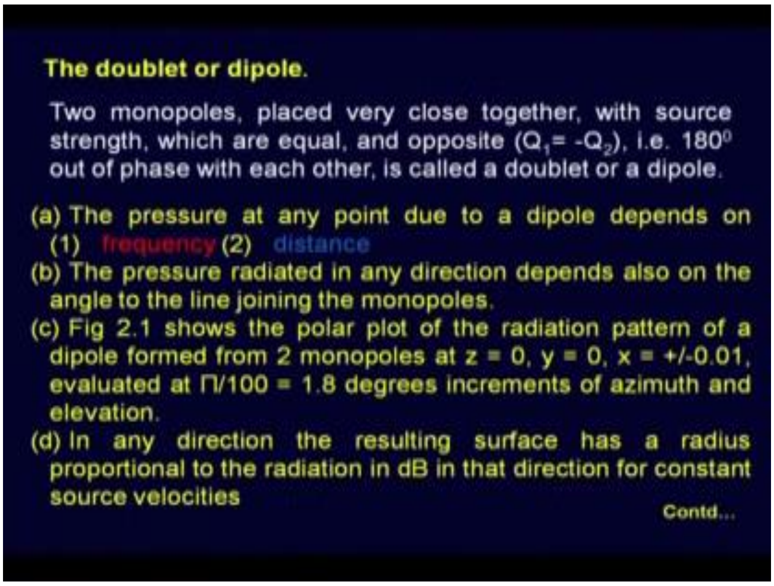

Now, dipole, what is dipole, if two monopole placed very close together with a source strength which is equal and opposite out of phase with each other is called the dipole. So, I required a two monopole placed in very close together with the source strength which are equal, but in opposite phase. This source and this source are equal, but in opposite phase, then I can create a dipole. So, pressure at any point due to a dipole depends on the frequency and distance. The pressure radiated in any direction depends also on the angle to the line joining the monopole. If the monopole are placed like this way this pressure, this point pressure and this point pressure will be different, because the monopole, the axis of the monopole if this then the radiation pattern will be look like this.

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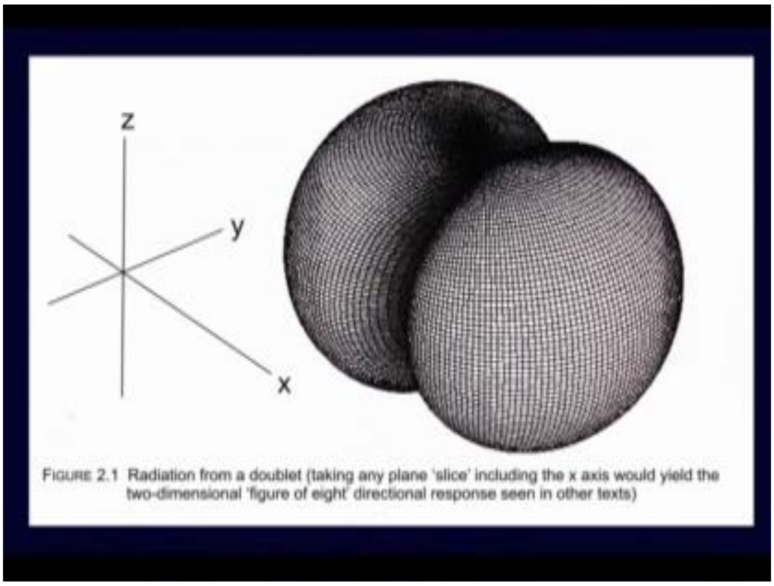

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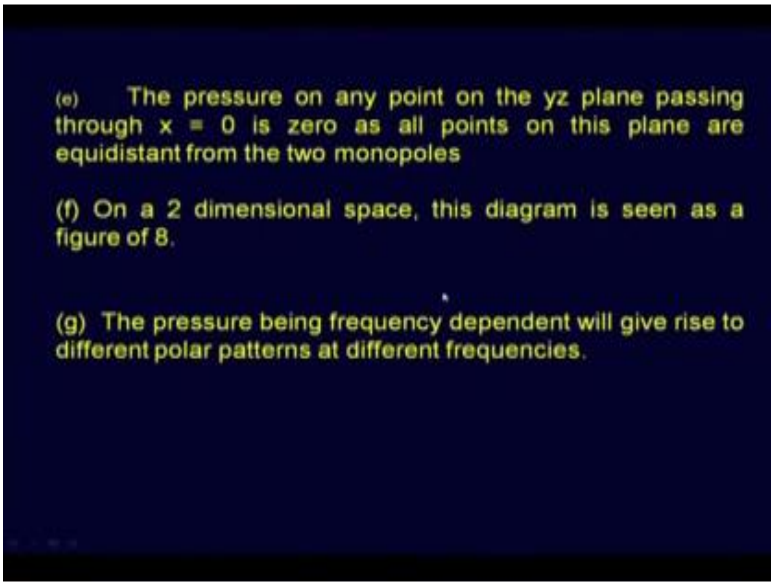

So, if you see the dipole radiation pattern this is the figure of merit of figure of eight for the dipole, I am not reading the slides. Now, I come for so you know what is monopole and what is dipole what is source strength.

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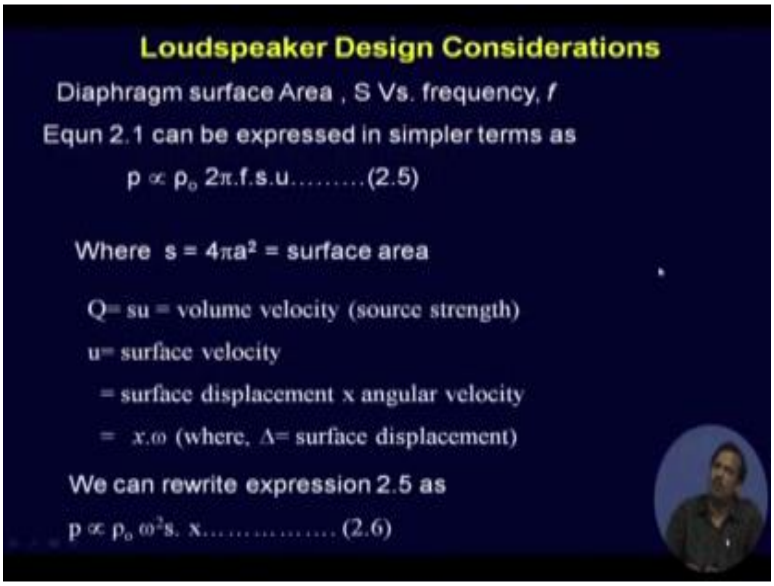

Now, loudspeaker design consideration.

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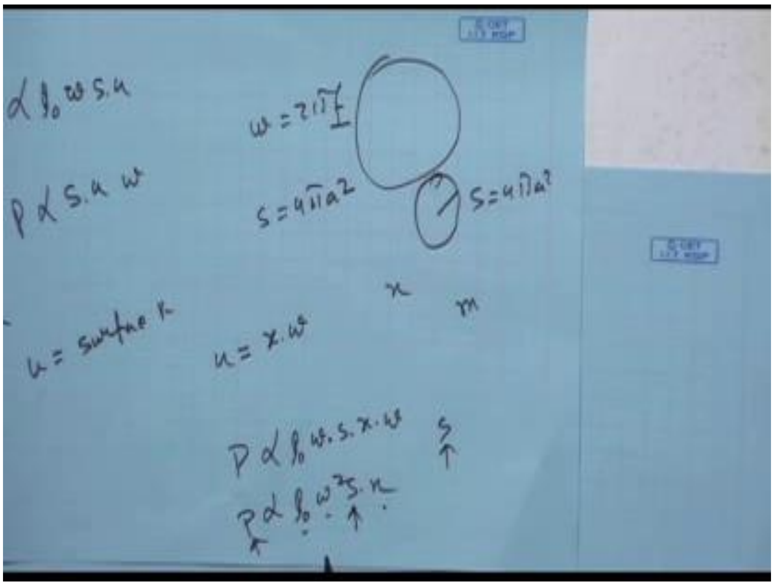

When you design a loudspeaker, diaphragm surface area if I change the surface area of the diaphragm then pressure is proportional to rho zero into omega into s into u by r is the distance where I get measure the pressure. So, I can say the pressure is proportional to the surface area and surface velocity, and frequency - omega is nothing but a 2 pi f. So, S is nothing but a 4 pi a square a is radius of the diaphragm and f is the frequency of the diaphragm. So, the surface strength of the loudspeaker is nothing but a s into u. So, it is nothing but a surface velocity. So, u is nothing but a surface velocity here, if you see u in here u is nothing but a surface velocity.

What is surface velocity u, surface velocity is nothing but displacements in angular moment, angular velocity multiply by the surface displacement, suppose it diaphragm is in here. So, what is surface velocity, displacement of the diaphragm multiply by the angular velocity? So, it is nothing but a x into omega, where x is the surface displacement. So, I can write p is proportional to rho 0 omega into s into x into omega. So, it is nothing but a rho 0 omega square s into x.

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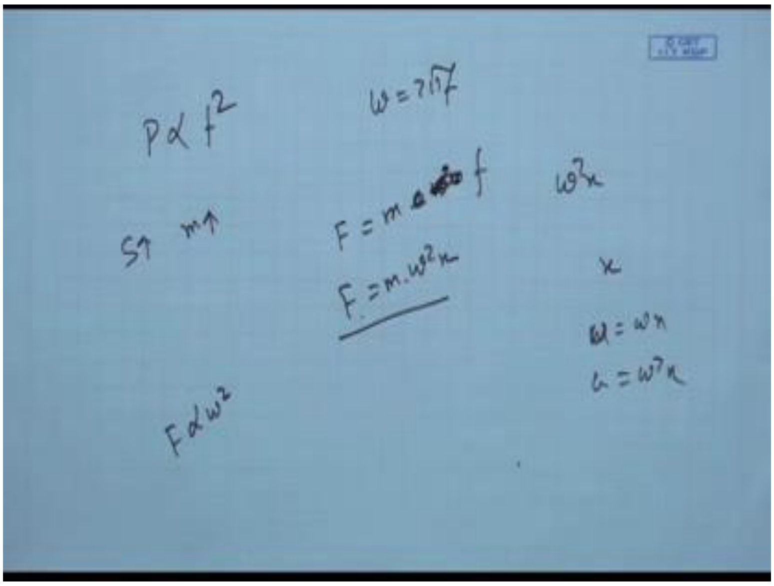

Or I can say that pressure produced from the source is proportional to f square where omega is nothing but a 2 pi f, so square of the frequency. So, P can increase pressure can be increased, if you see the P, P is nothing but a rho 0 omega square S into x. So, the sound pressure or acoustic pressure produced by the microphone can be increased by increasing the S, S is increased. Or if I increase S - surface area, what will happen, surface area that means if this is my diaphragm size, let this is my diaphragm size. So, surface area S is nothing but 4 pi a square, radius is a, if I change to this area, then diaphragm size is changed; if the diaphragm size is change, the diaphragm mass will be change. So, increasing surface area resulting that increasing mass of the diaphragm.

Now, if I say if the mass of the diaphragm is increased then the required force to move the diaphragm F is nothing but a mass of the diaphragm into acceleration. Let us acceleration is written by omega square; let this acceleration where mass into acceleration. So, let us acceleration is written by if F is not frequency correct force is equal to mass into acceleration. So, what is the acceleration, acceleration is nothing but a omega square x. If x is the displacement then u is nothing but omega x and u and a is nothing but omega square x. So, if it is that then I can write force is equal to mass into omega square into x, where omega square x is the acceleration of the diaphragm.

Now, F is proportional to omega square, the force required force to move the diaphragm is proportional to omega square. So, see if I want to produce a larger pressure acoustic pressure, I have to increase the area of the diaphragm. If I increase the diaphragm area, the mass of the diaphragm is increases and to move that mass I required more force. So, the electrical energy which will be converted to the mechanical energy should be high enough or produce that force which can move the diaphragm. Now, the force is proportional to omega square. So, if I change the frequency, so if I say I want to produce a 20 hertz or lets 30 hertz acoustic signal then it I can take a large diaphragm force is reasonable, but suppose I want to produce a 5 kilo hertz signal within a that same speaker then the force will be huge. So, my amplifier may not be able to deliver that force on that frequency to the diaphragm to move then diaphragm cannot produce that frequency.

So, what are the solutions? If you see if I there is a bottle neck in here, if I want to produce a large acoustical energy then I require the diaphragm area must be very high. Now, if I change the diaphragm area is high then the mass of the diaphragm is increases and I required a more mechanical force to move the diaphragm, which will come from the electrical energy. So, if you see in practical, there is a large box which can produce a large acoustic power by using the large area diaphragm area is very large. If the diaphragm area is very large, it can produce a large acoustics power is ok, but if the high frequency is has to be produced, then I required a more electrical power to move the diaphragm that may not be possible.

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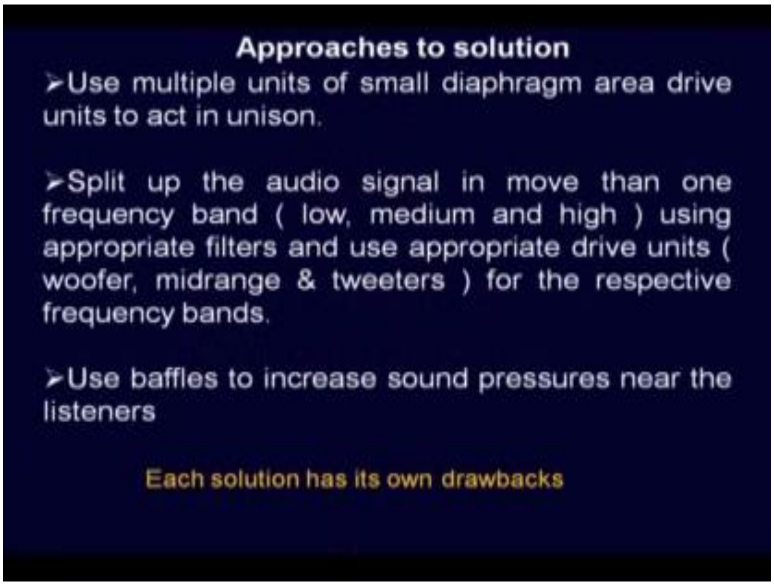

So, large acoustical energy when I want then if I want a large diaphragm maybe the high frequency component or frequency response of the loudspeaker maybe dropped or high frequency component maybe cut out. So, what is the solution, solution is that use multiple unit of small diaphragm. So, I want to increase the acoustic power.

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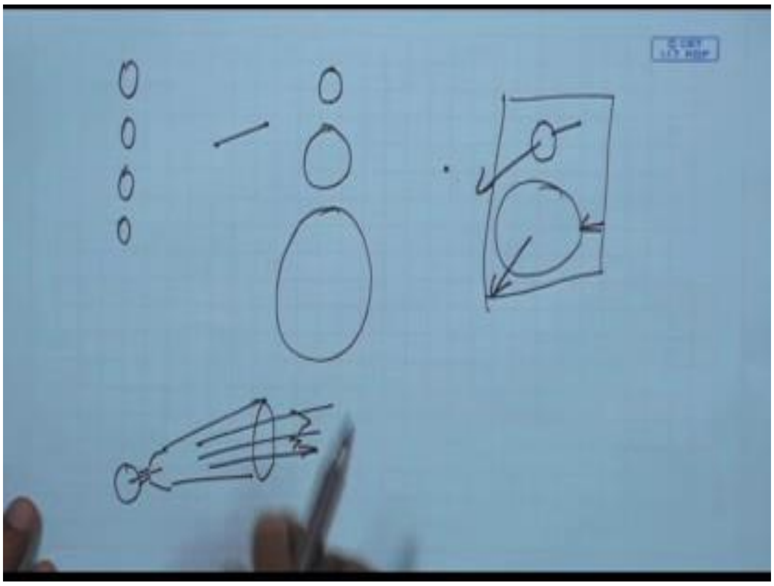

So, I can put several microphone together is one solution. Another solution is that I can split the audio signal whatever I want to feed and I can produce the high frequency audio signal by a smaller area of the diaphragm, and the low frequency mid frequency diaphragm a moderately larger area and a low frequency moderately high area of the diaphragm. So, the low is called woofer, medium is called midrange and high frequency loudspeaker is called tweeter.

In practical case, if you see any loudspeaker large loudspeaker system or large box if you see there is a two loudspeaker is mounted; one is huge and another is very small generally two are there. This is called tweeter and this is called woofer. So, woofer produces the low frequency sound and tweeter produce the high frequency sound. Now, you can say the low frequency sound will be high energy and the high frequency side will be low energy, yes. As per human listening system, we are much if low frequency with perceive the low frequency sound, we required much high amplitude; to perceive the high frequency, sound we required very low amplitude. So, our ear response is like that and also the frequency perception in human being is not linear; it is a non-linear scale. Lower side frequency with perceive very good, but when it goes to the upper side you may not perceive that good frequency difference.

So, what I will do since that my construction does not allow me to produce high frequency at larger power. So, I put a two loudspeaker in same system, one is produced low frequency and another is produced high frequency. So, the acoustical signal has the both the signal, but lower one is boosted up and upper one is by produce by the tweeter, so that is why the tweeter is important. If you cut down the tweeter then high frequency response, so the loudspeaker will be drastically change. So, if you say that ok, I want to use for a bias purpose maybe tweeter is not that much of equal. But if you want to listen a classical music on a loudspeaker, you may require a tweeter. And also that if you want to listen the classical musically huge loudspeaker it not that much of very good, if you want to listen a in a very good small loudspeaker. So, this is the reason why the tweeter is there in a loudspeaker system.

So, each and also I can produce the large sound using baffles. So, in generally, if you see in practical scenario that if you see the horn, have you seen the horn, there is a horn, mic and I put the here the loudspeaker is in here and it is connected to a horn. Why it is produced because the sound energy is produced in here it provides some directivity of the sound energy in single direction that is why the sound energy nothing but a mechanically amplified so that is called baffles. Baffles can increase the sound pressure reasonably high. So, if I put a loudspeaker if you see the loudspeaker has casing that is a box black box is there and then we house the loudspeaker in there, so that the black box is called baffle which increase the sound energy in that direction.

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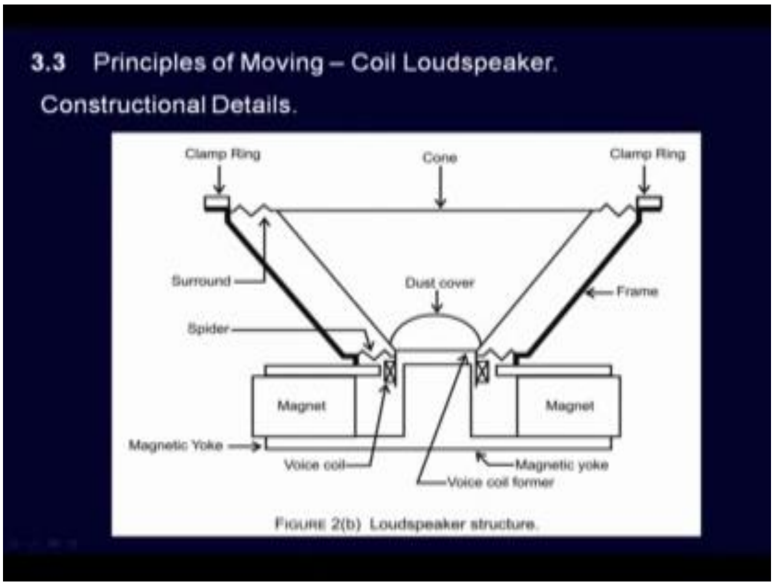

Now, I am not going this is the principle of moving coil loudspeaker constructional. This is same as micro phone construction because it is anti reciprocal transducer, it can act as a loudspeaker as well as it can access a microphone also, but in loudspeaker we make a very huge cone for the low frequency loudspeaker and for small low high frequency loudspeaker the cone size will be very small.

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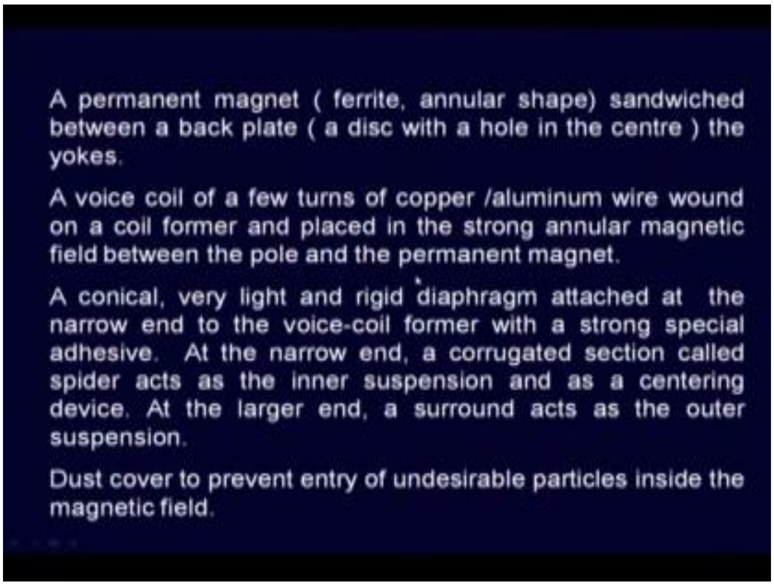

Then this is the arrangement you can read it and I will share the slide you can read it from there.

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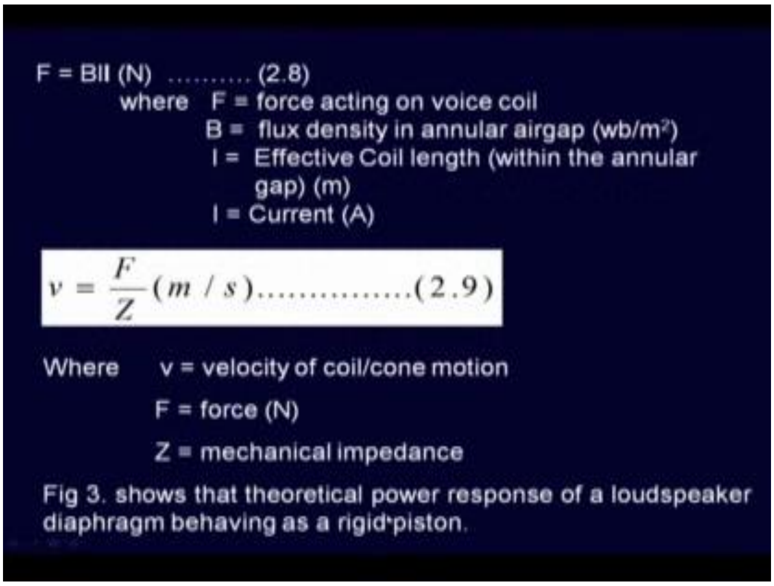

So, now there is a equation frequency response of the loudspeaker.

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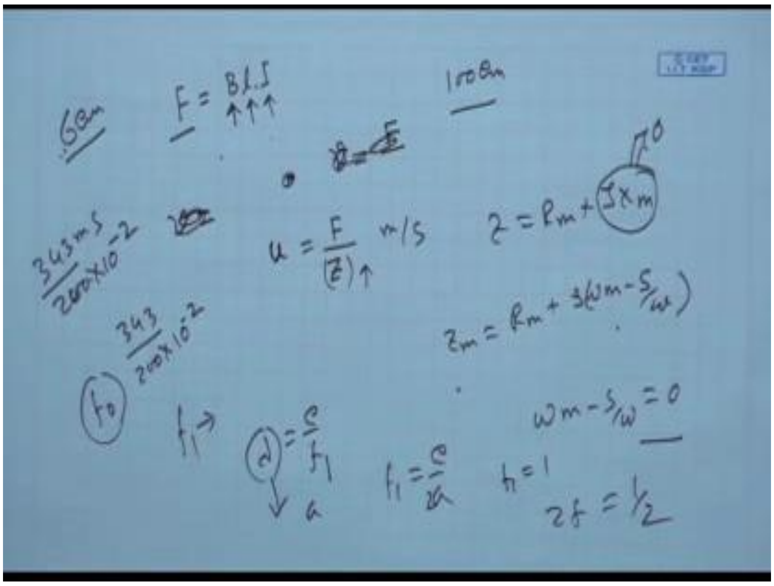

So, force produced by the loudspeaker F mechanical force is nothing but B into l into I, magnetic flux B is the strength of the magnetic spill, l is the length of the wire, I is the current pass through the wire plus density air gap force acting on a voice call effective coil length and current A. So that means if I have designed a moving coil loudspeaker and if I apply a or maximum loudspeaker or moving coil loudspeaker, if I apply a electrical current I and the length of the coil is l then the force acting on the diaphragm f will be B l into I.

Now, what is the velocity of the cone or what is the velocity of the diaphragm motion v or let us velocity write c, c is the sound velocity. So, write v velocity will be nothing but a force by or u velocity or let us write u is nothing but force by impedance, force by impedance meter per second. Now, what is impedance is a mechanical impedance force by mechanical impedance.

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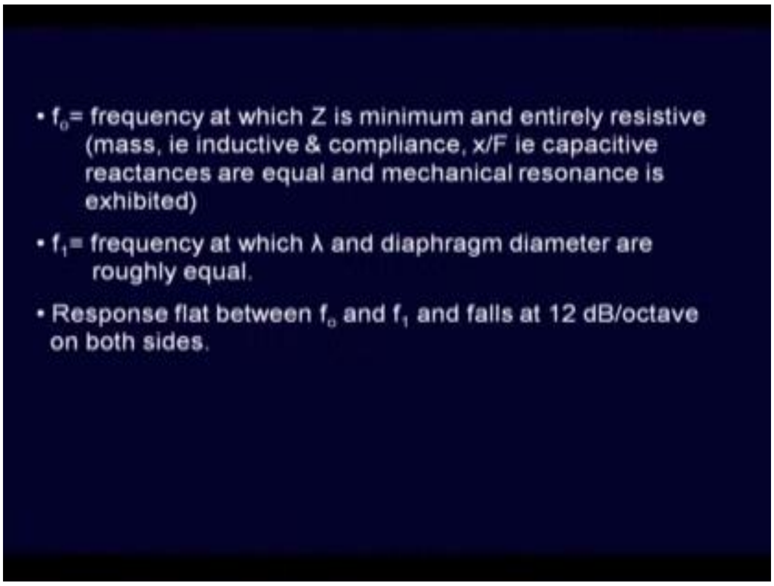

Now, if you remember the mechanical resonance then I said the Z is minimum this when the velocity will be maximum when Z is minimum. The Z is minimum when if Z is nothing, but R m plus J x m when the reactive component is vanish is equal to zero. So, only resistive then Z will be minimum. So, if Z m is equal to R m plus J omega m minus S by omega then at which frequency the Z will be Z m will be minimum when omega m minus S by omega is equal to 0. So, if x m is 0, z m is minimum then I can get the lower frequency, frequency response of the loudspeaker will like this.

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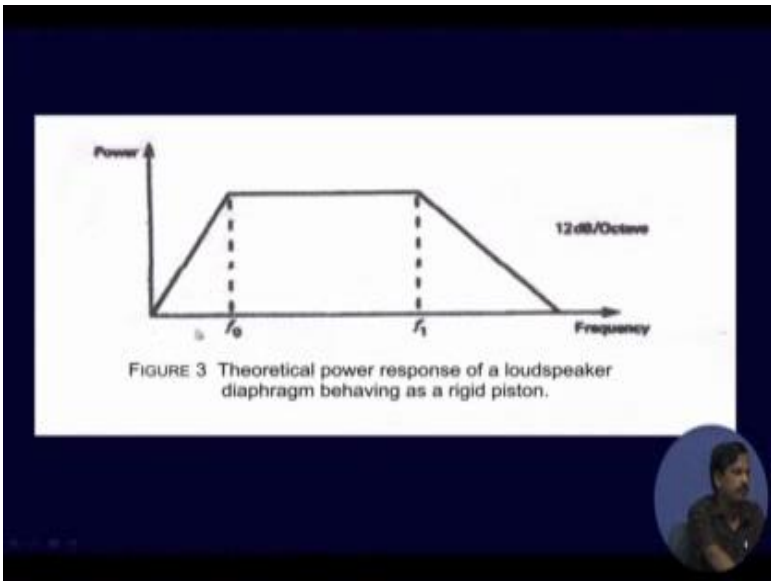

This is the power, this is the f 0 and this is the f 1. So, power produced by the loudspeaker will be flat during the f 0 to f 1. How to determine f 0, f 0 is a frequency at which Z is minimum that means entirely resistive no reactance. And f 1 is the frequency higher frequency this f 1 is the frequency where at which the lambda and diaphragm diameter are roughly equal the f 1 is that f 1 or what is the lambda for f 1 the lambda is nothing but a c by f. So, c by f 1 is the highest lambda. From this lambda is equal to the diaphragm area of the diaphragm then I call that is f 1. So, f 1 is nothing but c by a. So, diaphragm diameters are equal. So, the diameter means not radius 2 a, c by 2 a.

So, if I have a 343 meter per second is my sound velocity and if the diaphragm area is the if the diaphragm area let us radius is let us I can say 6 centimetre, not 6 centimetres, 6 inch or let say I said 100 centimetre diaphragm, 100 centimetre radius diaphragm. So, it is c by 200 centimetres. So, it will be 10 to the power minus 2. So, it is nothing but 343 divided by 200 into 10 to the power minus 4. So, high the radius high diameter loudspeaker cannot produce the flat freq flat cannot deliver the flat power at a high frequency that is why I require the tweeter. And it will be degrade by 12 dB power of ten on both side, every doubling the frequency it will be degrade by 12 dB. So, at twice f 1 if f 1 is equal to 1 twice f one it will be half power.

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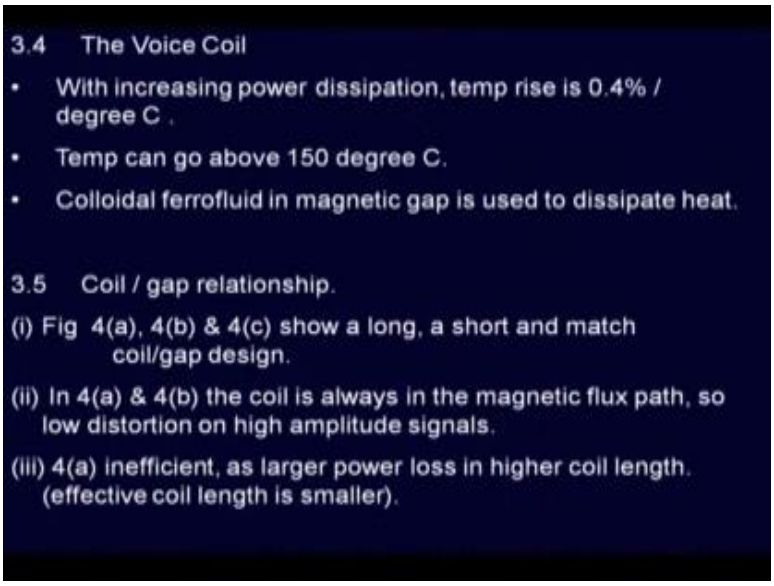

Then the voice coil. What is the voice coil with increase the power dissipation temperature will be rise then temperature can go up to 150 degree centigrade. So, how do reduce that things loss by coil gap the coil gap relationship.

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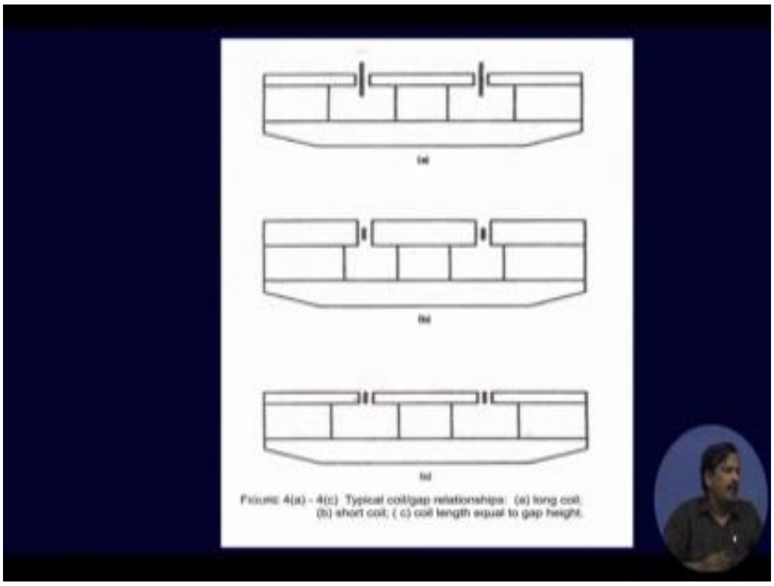

So, I can construct the loudspeaker like this, the coil is this is the magnetic gap and coil can be designed like this way. So, there will be a loss and this is the magnetic field coil can be designed for this portion only or I can design the coil equal to this magnetic area. So, this all three has an advantage and disadvantage. So, if it is there, so the long short match coil. So, coil is always in the magnetic flux path, so the low distortion on high amplitude signal a.

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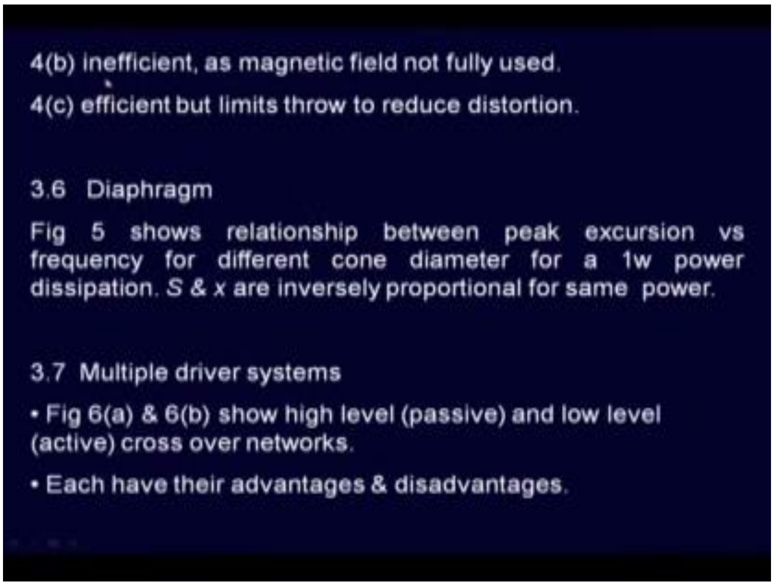

And b the coil is always in magnetic path and c, a is inefficient as a larger power loss in higher coil length. And if you see b is inefficient as magnetic field not fully used and efficient, but limit throw to reduce distortion. So, these are the - a, b, c picture. So, this we can read and that. Then diaphragm then multiple drives systems. So, how do you what I said that if I increase the area of the diaphragm then that it cannot produce the high frequency.

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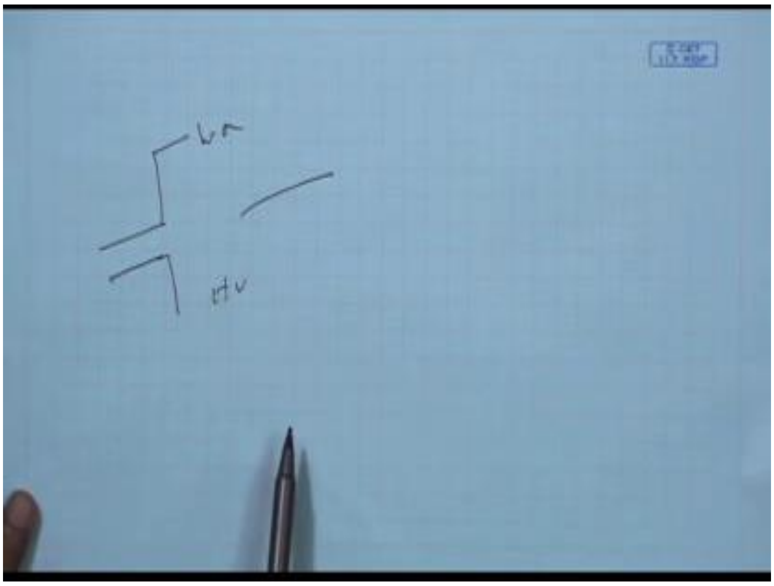

So, for the high frequency, I can divide at the input signal in two paths - two area one is the low frequency and another one is the high frequency then I can drive the two loudspeaker. So, signal has to be split tweeter is the high frequency and that other one is the low frequency.

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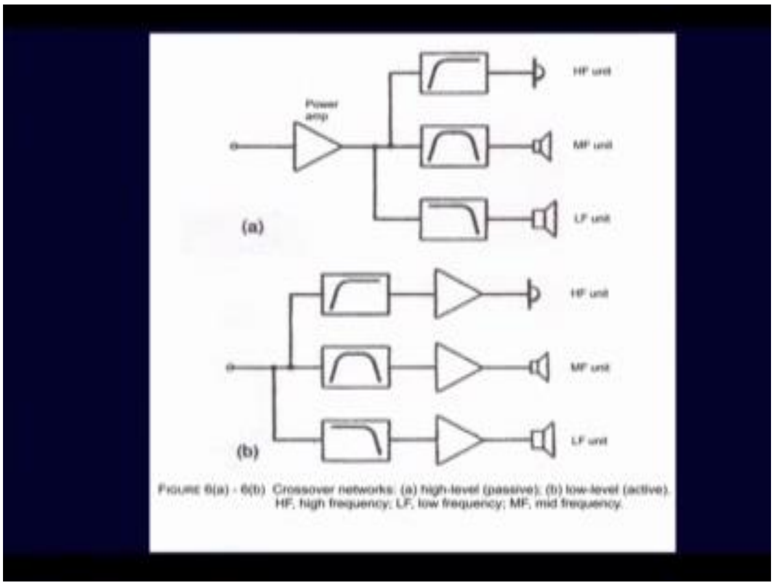

Now, I will gradually I can this is the not required you can go through the slides.

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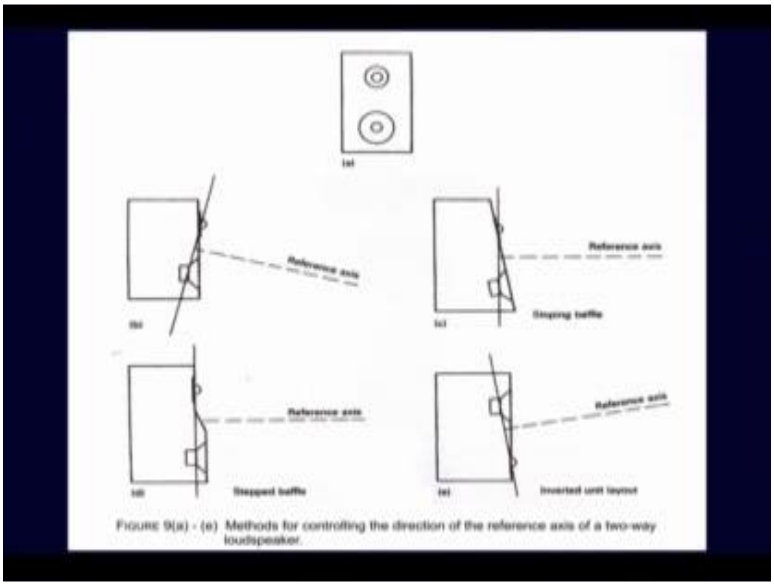

Then I show you something on reference axis loudspeaker housing is not that easy. So, if it is this tweeter and main loudspeaker in this the same plane then the same reference axis is this one. If it is housing like this then the reference axis is this one. So, radiation power will be in this direction and this reference axis is this if I put like this to cope this is the way reference axis is this. So, different construction gives you the different reference axis.

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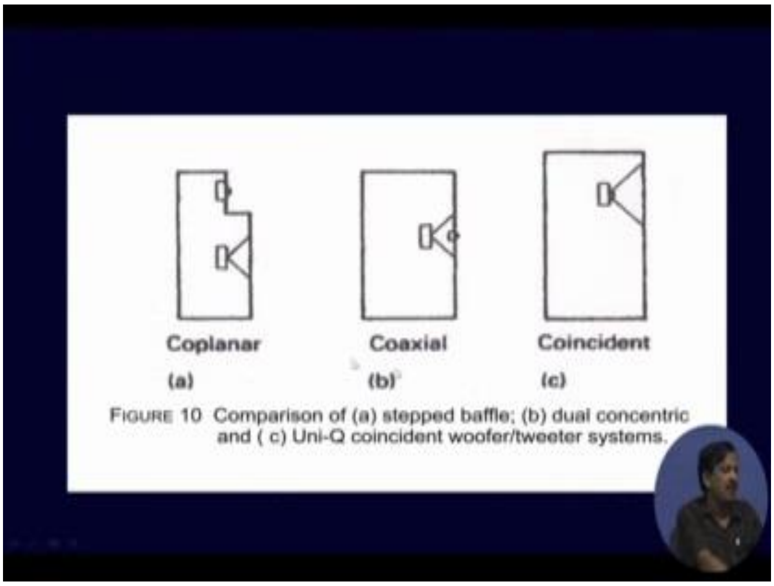

And there is another thing is the baffle design of the baffle, coplanar, coaxial, and coincident.

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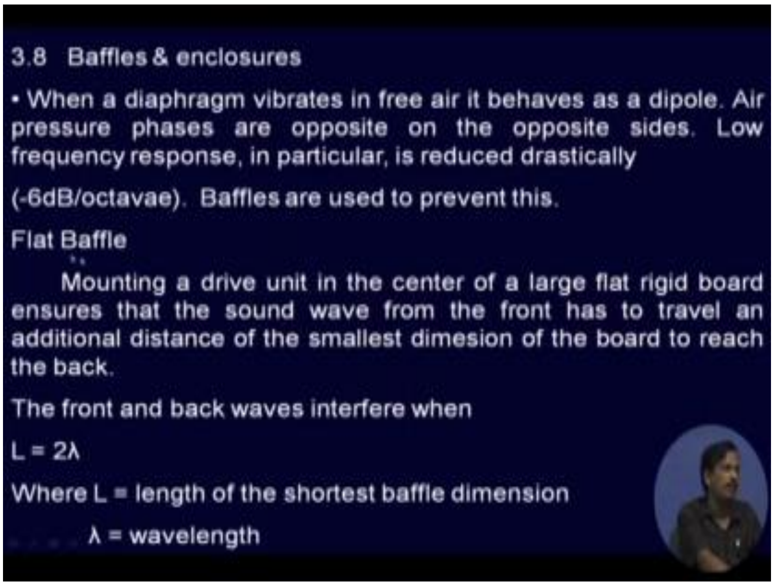

Then there is baffles design baffles and enclosures are also very important for loudspeaker. So, you can read these slides, because there is lot lesser times.

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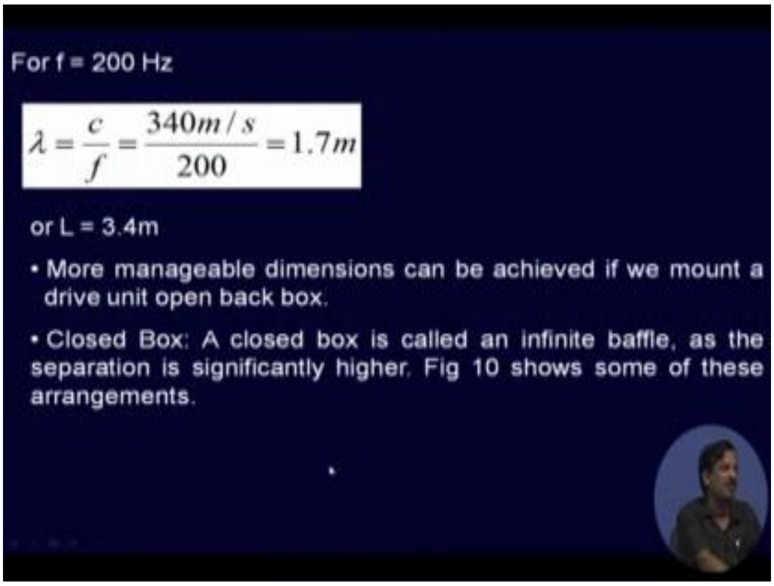

So, I cannot go details on the baffles and enclosure roughly it is nothing but a 3.5 meters.

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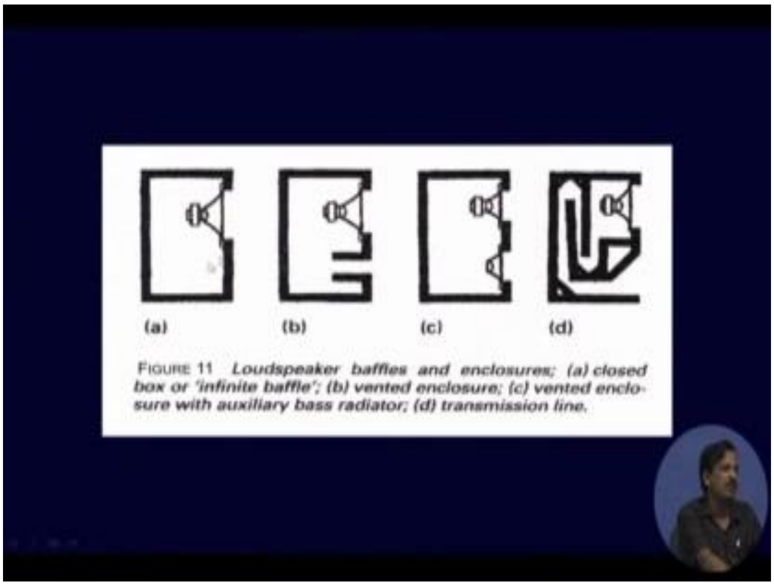

And it can be reduced by constructing the black box and then I housing the loudspeaker. So, this is the more or less loudspeaker design. Main part is that how do you decide the area of the diaphragm, and you have to design you have to find out the f 0 and f 1, and then the mathematical calculation will come from the reciprocal and in the reciprocal constructional design.

Thank you.