

Architectural Acoustics
Prof. Shankha Pratim Bhattacharya
Department of Architecture and Regional Planning
Indian Institute of Technology, Kharagpur

Lecture – 26
Electro – Acoustics – I

Good morning, student, welcome to the NPTEL course on Architectural Acoustics today we will going to start the sixth week program of the total 8 week module. So, lecture number 26, today will deal with the Electro- and I have two lectures for that, the lecture 27 is also on the Electric Acoustics.

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Learning Objective

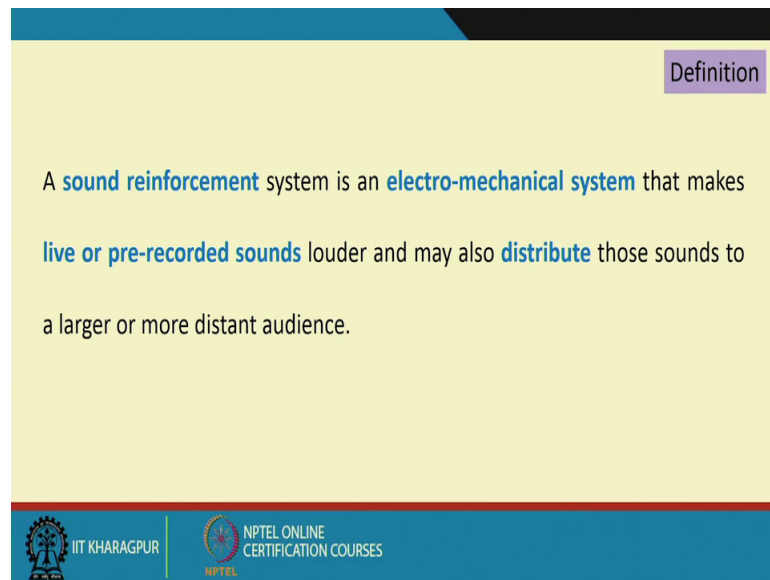
Discuss the various component of electro-acoustics

Relate the fundamentals of electro-acoustical parameters to design

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So, let us first go to the learning objective of this 26th lecture. The electro-acoustic is a one of the integral part of the acoustics or modern days acoustics I must say and the electro acoustical component it has actually various components and from the various component there are various fundamentals and the some properties of the those component. In the lecture number 26 we will discuss the various component of the electro-acoustic that is first and then will try to relate the fundamentals of those principles or the parameters electro acoustical parameters and try to relate that with the architectural design of electro acoustics.

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Definition

A **sound reinforcement** system is an **electro-mechanical system** that makes **live or pre-recorded sounds** louder and may also **distribute** those sounds to a larger or more distant audience.

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Now, basically where I say about the electro-acoustic it is kind of a sound reinforce system. The sound reinforce system is sometimes when the a particular hall or particular area is too large or sometimes I am going to produce some kind of a pre recorded kind of a voice or music we need to help we are need to take help of the electro mechanical systems, something like a loud speaker the microphones and all.

In fact, today this NPTEL course or may be this NPTEL course and all everything is actually a the game of a electro acoustical part. I have a speaker with I mean the microphone with here which is a collar microphone kind of a thing and that is going to produce the sound to you through some electro electrical pulse and then finally, my sound is transmitted or the captured by this particular this collar microphone and it is sending some kind of electrical pulse and finally, when you reach into your the environment or the your room or in your laptop or may be any kind of a computer device or maybe it is your mobile. It is actually again translate that electro I mean electrical pulse to a sound energies also. So, this is the brief definition of the sound reinforcement or the sound electro acoustical systems.

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Objectives

- To **reinforce** the sound, which would otherwise be inadequate.
- To provide **adequate** loudness and intelligibility.
- To **reproduce** the sound, which was recorded earlier

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Now, what are the objective of this electro acoustical systems? There are three basic objective of the electro acoustical systems, one is reinforce the sound. I just I have told you if the size of the this a vertical hall is very large or there is a huge amount of audience very high capacity of the volume I have to take care, sound probably in probably it will be in adequate I mean the only the voice or any kind of a music which actually going to play in the stage which is may be very weak for the rear seat are some portion of the auditorium. So, you need to reinforce the sound by some kind of a amplification.

Then the we have to provide some kind of the adequate loudness or may be the adequate intelligibility where we can actually hear the sound actual sound and the frequency of the sound without any kind of a distortion and the loudness should be adequate. And, sometimes we need many times we need the reproduce the sound which was recorded earlier some music or maybe some kind of a the voice or some kind of the performance which has been recorded earlier and that that can be reproduced in a some different locations. So, for that electro acoustical system is required.

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The slide is titled "Requirement" in a purple box at the top right. It lists five requirements in a list format:

- Travel to the audience at larger distance
- Adequacy of sound level
- Uniform sound level
- No distorted frequency of sound
- Durability and economy

The slide footer includes the IIT Kharagpur logo and the text "NPTEL ONLINE CERTIFICATION COURSES" with the NPTEL logo. A small number "5" is visible in the bottom right corner of the slide.

Now, in this when we going to deal with some kind of electro acoustical systems various component of it we have some of the requirement. In requirement from that point requirement point of view of the first requirement the sound should travel to the audience of the in large distance, that is the one of the basic fundamental requirement of electro acoustical systems. Otherwise perhaps electro acoustical systems was not developed.

And, the second requirement is the adequacy of the sound level sound level has to maintain a kind of a adequate sound level in every part of the auditorium not only in the front or may be only in the back part, in the sides and also in the middle part of the auditorium. The uniform sound level which is also a part of are kind of the adequacy a point of view we can just the uniformity. So, there should not be very high sound level in some point and very low sound level in some other point. So, should have a flat kind of a character.

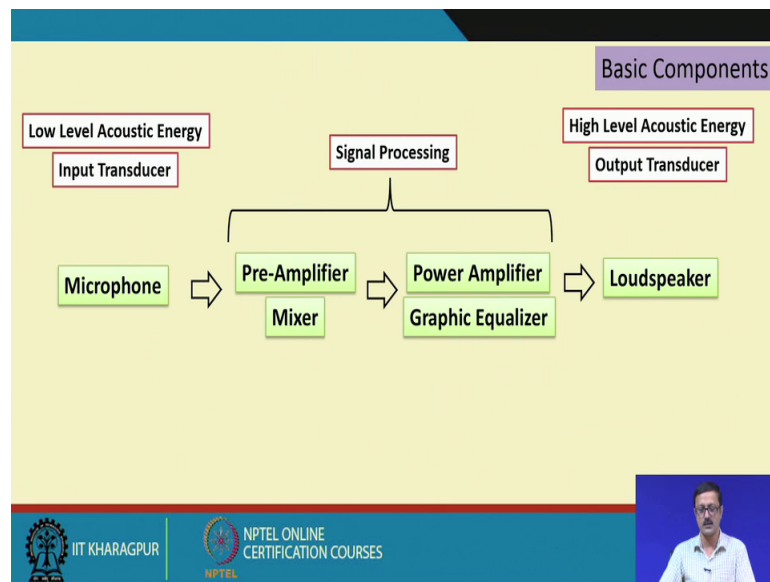
The fourth one is very important there should not be any distortion of the frequency of the sound ah. So, the sound frequency which comes out from any kind of a musical instrument which comes out from some kind some someone's voice should not be started the same frequency and as you know the frequency is actually give you the character of the sound. So, that frequency should not be distorted. Suppose, a tabla is playing over here through that sound should be a bass or those kind of a frequency level and may be a

guitar or may be any other kind of a instrument should not or may be a piano should not distorted the voice after this amplification or the reinforcement.

And, the finally, which always a architect deal or any kind of a engineer deal which is a system durability and the total economy of the systems that also we have to check from the various properties of the system component.

So, next we will see the what are the basic component of the system some this your acoustical systems.

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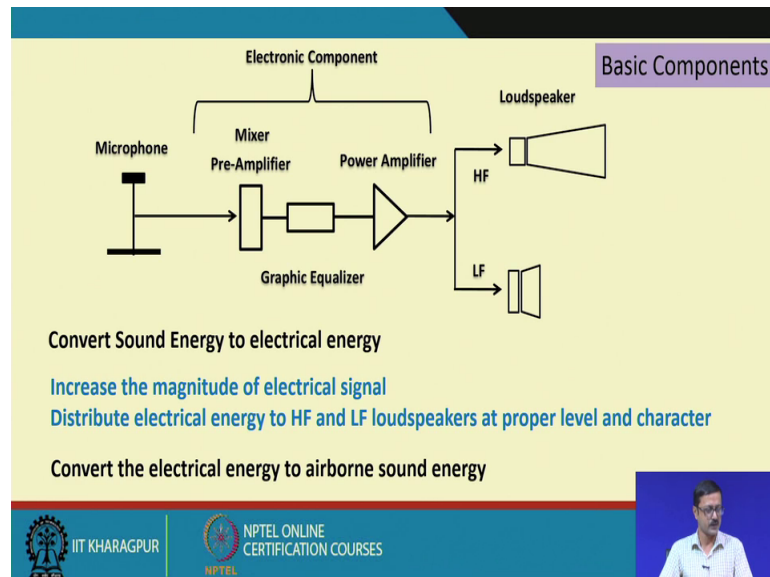


So, if I see in this particular slide I have classified this system into three major part. The initially it is a low level acoustical energy which is provided by some kind of a performance or may be some kind of a speech has to taken into some kind of a input transducer and then this particular thing the particular acoustical energy has to go through some kind of a processing which is called the signal processing and then the high level acoustical energy has to be reach to the audience through some kind of the output transducer.

And, this input transducer are the low level acoustical energy can be taken through the microphones which I have just now told you that this I have also a microphone only and then this particular microphone then going to the town from the microphone captured sound from the microphone when to the signal processing unit which has basically two

power; one called the pre amplifier or a mixer and another one is called the power amplifier or the graphic equalizer and the finally, it goes to the loudspeaker which is output transducer and the role of the loudspeaker is just translate the electrical signals to a sound signals once again.

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I have given this particular slide is some pictorial presentation. So, with that I can again re state that particular functions or the basic component. Microphone which convert the sound energy to electrical energy, then there are lot of electronics arrangement where the graphic equalizer, the power and the pre-amplifier has to make a common role which is comes under some kind of the signal processing kind of thing where they will actually separate out the frequencies HF and LF are the high frequency and the low frequency sound to the high frequency and the low frequency loudspeaker and finally, the loudspeaker will convert the electrical energy should to airborne sound energies also.

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Microphone

Microphone is an instrument that **convert sound waves** (such as speech or music) into **electrical energy** variations which may then be amplified, transmitted, or recorded.

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So, the first component if I again go back, so, the first component is this particular the microphone this microphone is the first component. So, let us discuss to one by one of this component and what are the various, the requirement and what are the various types available ah. The microphone again what we have a understand that is convert the sound energy or the sound wave to electrical impulse.

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Types of Microphone

Shape and Use	Directivity
Handheld Microphone	Omnidirectional
Shotgun Microphone	Bidirectional
Ribbon Microphone	Unidirectional
Condenser Microphone	
Dynamic Microphone	

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And the types of if I say the types of sound sorry the microphones or how I has divided into two type of that classification. In the first type of classification I can say it is some

kind of within shape and the use point of view there are classification like handheld, shotgun, the ribbon or condenser or dynamic kind of a microphone. And, if I went to some kind of a directivity point of view the how the microphone is behaving and how it is going to captured the sound and from which direction actually, then it has a omni-directional, bidirectional or unidirectional; this three categories of the microphones are available and that is the classification based on the directivity.

So, let us go one by one on see how this are actually look like.

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The slide is titled "Types of Microphone" in a purple box at the top right. It features two main sections. The first section, on the left, is titled "Handheld Microphone" in blue text, with the subtitle "Entertainment / Reporter" below it. It shows a silver handheld microphone with a red "NEWS NEWS" sign attached to its handle. The second section, on the right, is titled "Shotgun Microphone" in blue text, with the subtitle "extremely directional pickup pattern popular for TV news and movie sets." below it. It shows a long, black, cylindrical shotgun microphone. At the bottom left, there is a small text credit: "Photo source: <https://ehomerecordingstudio.com>". At the bottom of the slide, there are logos for "IIT KHARAGPUR" and "NPTEL ONLINE CERTIFICATION COURSES". A small inset video of a man speaking is visible in the bottom right corner of the slide.

And, the shape point of view and the use point of view this handheld kind of microphone which you must have seen in the hand of reporters, in some kind of entertainment channels, they put in front of some somebody's face and the gives some kind of a bites the that particular the journalist and they use this particular type of handheld microphone kind of a thing ah. There are two types of handheld; one is with cord or sometimes it is without cord are the cordless kind of a microphone are also available.

In the next one is a shotgun microphone and that is extremely directional microphone, but; that means, it is actually catches the sound from a very the distinct the direction point of view. So, if that particular direction has changed from some angle it has a very poor kind of a output, it will show some kind of a very poor output. It is very popular in the movie set and particularly the news televisions or news studios, TV a news studios

where a particular shotgun kind of a microphone is held in a particular direction or particular place and the speech or may be some kind of the dialogue can be captured.

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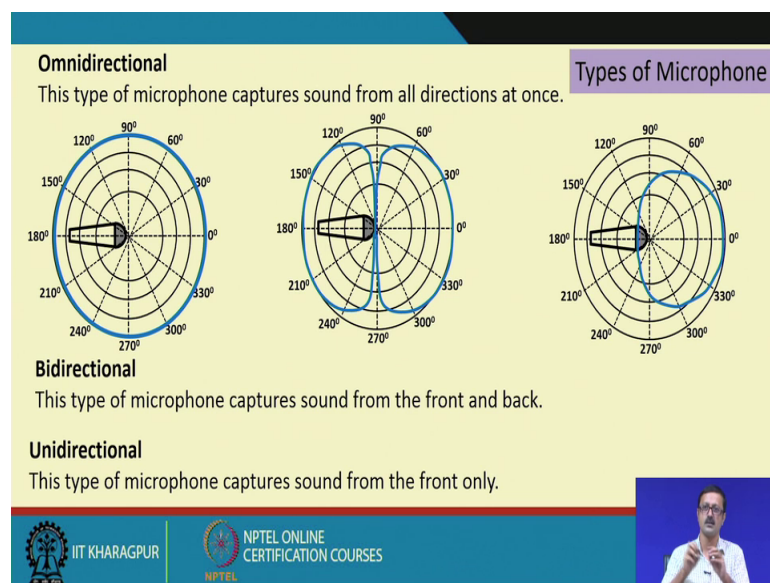
The next one is ribbon microphone which actually use for the speeches which actually use for the radio stations which this ribbon microphone are bit older version and this modern days we have some better versions of this particular microphone and it was used in kind of announcement or those kind of a thing. If few hap perhaps seen this kind of microphone in the telecasting when they actually telecast the videos for the some the sports event like cricket or may be football in that particular both where actually the commentators both where they actually sit and give the running commentary of any kind of a sports event they use this kind of microphone. Radio stations also for some kind of announcement in the railway stations or may be any airport they use this kind of microphone, ribbon microphone.

Then the condenser microphone again it is an old kind of version of the condenser microphone which is a very wide range of the directivity and this actually captures the vocals and the music of different area if must of we have seen in a stage performance musical performance like in a stage, in auditorium this condenser microphone are each given to far for the each the vocal or the musical instrument and it captured the sound from them.

The last one is called the dynamic microphone which is mostly used for the instrument like the guitars or drums again this has a wonderful directivity and this is a kind of a replacement of the condenser microphone to this dynamic microphone nowadays and the again I must say this dynamic microphone is the that what we have seen the very first that is the that one handheld is one type of the dynamic microphone kind of it. So, those are the various classifications of the microphone from the virtue of the shape and the use.

So, let us see the what are the types available from the directivity.

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So, the first one is the omnidirectional, where I have seen I have plotted a diagram with a center point I place that particular microphone and this blue circular line gives the directivity it has. So, it shows that it has almost the equal directivity which is the and it captures the sound from 360 degrees. So, that is why it is called the omnidirectional kind of a microphone. And, the bidirectional microphones are those which have a directivity only in the two opposite directions. So, one may be in the 0 degree direction and another one is in the 180 degree direction.

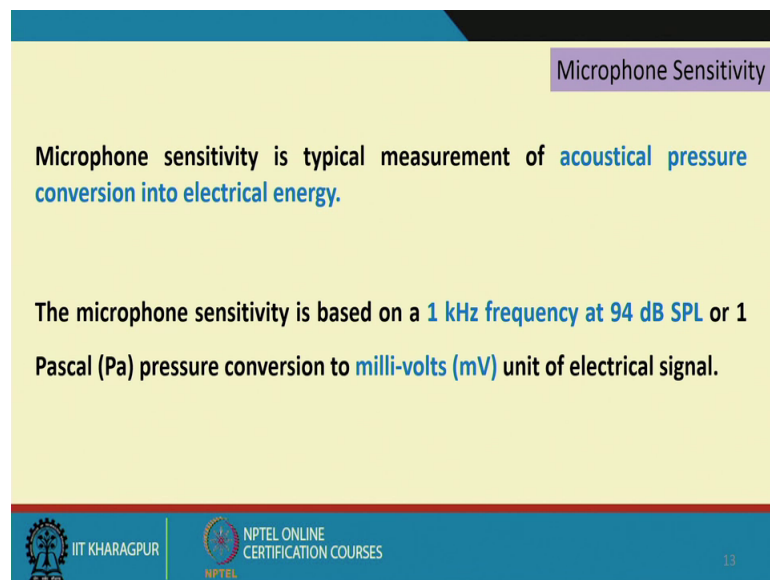
So, if you see the blue lines over here in this particular diagram. So, from that you can see, the offset from the towards the 90 degree and the 270 degree the capturing potential is little less. So, it is the property of this bidirectional kind of a microphone. The third one is the unidirectional type of microphone where it actually captures the sound only from the one location only from the front location. So, if it is in the 90 degree orientation

are 180 or may be in the 270 degree orientation some sound comes from this side or may be from back side or from these area it does not have that much of a fifty bit.

So, sometimes we use the unidirectional microphone just to avoid some kind of noise maybe background noise or may be noise from the sides. So, only when you focus a particular the source of the sound and you know that particular source of the sound is located it the particular the typical point, then it is better to the unidirectional kind of a microphone.

Now, next talk about some kind of the what are the different properties of the this microphone.

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Microphone Sensitivity

Microphone sensitivity is typical measurement of **acoustical pressure conversion into electrical energy**.

The microphone sensitivity is based on a **1 kHz frequency at 94 dB SPL** or **1 Pascal (Pa)** pressure conversion to **milli-volts (mV)** unit of electrical signal.

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The first property of the microphone is called the a sensitivity of the microphone how the how thus microphone is sensitive how much it sensitive. So, it is relate it actually translate in a the definition wise that how much acoustical pressure is converted to this particular by this particular microphone some kind of a electrical impulse because I have already told you, what is the duty on the role of a microphone it is captured the sound and then it translate the sound that particular a mechanical the vibration of the sound to a electrical impulse.

So, the one kilohertz frequency of the sound at 94 dB SPL or 1 Pascal pressure sound pressure is the bottom or the decome level and from that how much it is converting in

millivolt unit in electrical signal is sensitivity of a microphone. So, some microphone may provide much more millivolt and some microphone may not provide that much from that point of view I can give some kind of a stratification of the microphone sensitivity, but the input is 1 kilohertz with 94 dB sound which is equivalent to almost 1 Pascal.

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Microphone Sensitivity

Sound Input: 1kHz, 94 dB = 1 Pa

Electrical Signal Output in mV

$$20 \log \frac{P}{P_{ref}} = 94 \text{ dB} \Rightarrow \frac{P}{P_{ref}} = 10^{\frac{94}{20}} = 10^{4.7}$$

↓

$$\frac{P}{P_{ref}} = P_{ref} \times 10^{4.7} = 2 \times 10^{-5} \times 10^{4.7} = 1.002 \text{ Pa}$$

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So, I have given a pictorial diagram. So, suppose this is the audio wave or the sound wave which is actually mechanical wave it is the compression read of fraction it is come to this particular microphone. So, the input has to be 94 dB at 1 kilohertz and that equivalent to 1 Pascal. So, I have just calculated that the this formula which 94 dB is the it is the pressure, I want to find out it is 20 log and then if I just rewrite this equation then I find that this P which is equivalent to 94 are which produce the 94 dB is nothing, but are one almost 1 Pascal. So, a 1 Pascal sound pressure is generated very near to the microphone and you find out how much is the millivolt output from some kind of the electrical impulse recorder you find out what is the millivolt output and that will be the sensitivity of the microphone.

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Signal Processing

Pre-Amplifier

Mixer

Pre-amplifier **amplifies low-level Signals**

It also **combines and select the signals** as required (Mixer)

Pre-amplifier **minimise the noise** that may present in the signal

A pre-amplifier **processes a signal** to make it fit **for the next stage** in the signal chain.

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After the microphone the next stage what so, what microphone does is that it has actually converted it actually converted the signal which is sound or the mechanical signal to a is acoustical signal to a electrical impulse. Now, it will go to the some signal processing which is the secondary are the middle unit of my the electro acoustical systems. So, it has first has to go through the preamplifier which is also called as any mixer.

So, it is mix, first of all it is going to mix some kind of frequencies combined some kind of frequencies why because there may be lot of the vary I am lot of the more than one output. Suppose, there are some mis instrument some vocals and some other people may give some kind of a speech. So, in a stage or in a music recording room there may be different musical instrument. So, you captured different the sound are the electrical impulse by virtue of different microphones and then there is a very low level of signal amplification very slightly low level of signal amplification it will do.

Then, in this particular process of the preamplifier it will also minimize the noise if there are some kind of a noise which is not wanted for the final delivery. So, those noise has to be minimized by virtue of some kind of a screening and filtering in this particular preamplifier system.

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The slide is titled "Signal Processing" in a purple box at the top right. It features a block diagram with four green boxes: "Pre-Amplifier" and "Mixer" are on the left, "Power Amplifier" and "Graphic Equalizer" are on the right, and a white arrow points from the left group to the right group. Below the diagram, there are two paragraphs of text. The first paragraph defines a Power Amplifier, and the second paragraph defines a Graphic Equalizer. At the bottom, there are logos for IIT KHARAGPUR and NPTEL ONLINE CERTIFICATION COURSES.

Pre-Amplifier
Mixer

Power Amplifier
Graphic Equalizer

A **Power Amplifier** is an electronic device that can **increase the power** of a signal by increase the amplitude of a signal

A **Graphic Equalizer** is a high order **audio control** unit that control a number of different **frequency bands** at a specific audio range.

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And, this particular mixing noise minimization and the amplification slight amplification I must say that will be given to the next part of the signal processing which is called the power amplifier or it is also known as sometimes graphics graphic equalizer.

Now, they it has it has major majorly two jobs. This power amplifier power amplifier as the name suggest it will amplify the power amplify the increase the power of the signal and then it will which is one of the need for the electrical sorry the electro acoustical systems which is going to reinforce my sound and the graphic equalizer controls the audio, total audio and different frequency bands and everything they will give you kind of a overall control by virtue of some kind of you want to produce some kind of a music suppose you want to put the tabla sound little low and the guitar sound little high and the voice of a particular singer is the further more high or low whatever may be the situation or a director music director thinks, by virtue of that particular thinking we can use some kind of the frequency band control and the equalization of this particular sound.

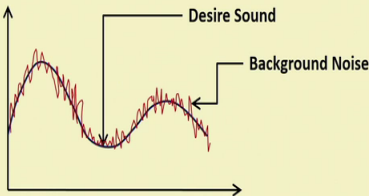
So, that is one of the part one of the role or one of the component that will actually perform in the power amplification.




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SNR

Signal-to-Noise Ratio (SNR or S/R)

Signal – to – Noise Ratio is defined as the ratio of signal power to the noise power.
 The ratio is usually measured in decibels (dB) using a signal-to-noise ratio formula.
 It compares the level of a desired sound signal to the background noise level.

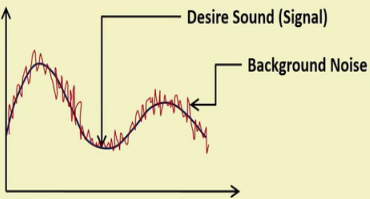


So, after this power amplification was going on so, there is a kind of a another ratio comes or another parameter comes which is called a signal-to-noise ratio because there are some noise which I have shown here by virtue of some blue sorry the red lines which is a noise, but desires sound is this path that has to be taken into accounts. So, I have to actually take out this particular noise and then the desire some has to filtered.

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SNR






Sound Power	Sound Amplitude
W_{Signal}	A_{Signal}
W_{Noise}	A_{Noise}

$$\text{SNR}_p = \frac{W_{\text{Signal}}}{W_{\text{Noise}}} = \left(\frac{A_{\text{Signal}}}{A_{\text{Noise}}} \right)^2$$

$\text{SNR}_p = 1 \dots\dots\dots \text{SNR}_{\text{dB}} = 0$

$\text{SNR}_p = 100 \dots\dots\dots \text{SNR}_{\text{dB}} = 20$

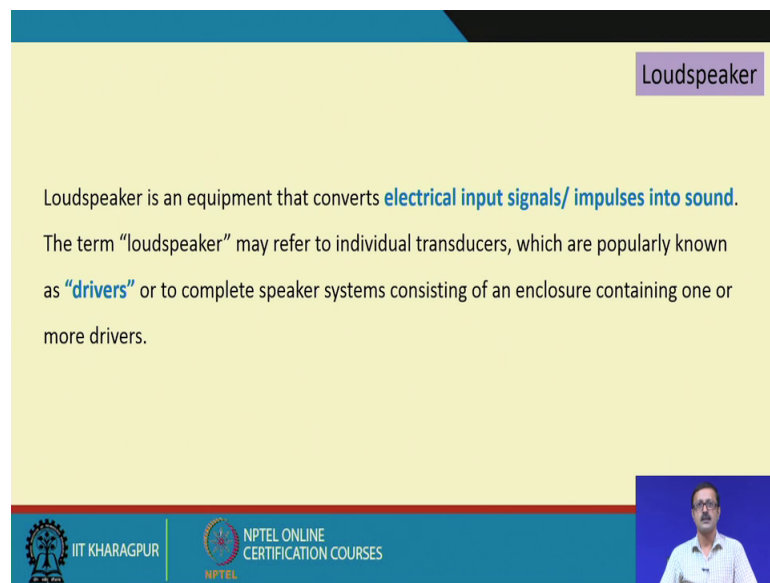
$$\text{SNR}_{\text{dB}} = 10 \log \left(\frac{W_{\text{Signal}}}{W_{\text{Noise}}} \right) = 20 \log \left(\frac{A_{\text{Signal}}}{A_{\text{Noise}}} \right)$$

At for that the signal-to-noise ratio has been the taken as a one of the basic parameter where it has been right as from the power point of view W signal by W noise, where the

When we are talking about the power of this particular signal and this noise or sometimes it may be rewritten as a dB scale when it is SNR is a signal to noise ratio in dB which is $10 \log$ of that on it can be also rewritten as a amplitude functions also when it will come on to I mean the it is amplitude square is the way it can be written. So, I can say that when the from the power point of view SN ratio is 1 so, its translate as a dB of SN ratio of 0 and when it is suppose 100 from power point of view it is goes like 20.

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Loudspeaker

Loudspeaker is an equipment that converts **electrical input signals/ impulses into sound**.

The term "loudspeaker" may refer to individual transducers, which are popularly known as "**drivers**" or to complete speaker systems consisting of an enclosure containing one or more drivers.

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(A small video inset of the presenter is visible in the bottom right corner of the slide.)

So, after this the signal processings and all we will let us let us see what are the loudspeaker. Loudspeaker is the final the final part of this electro acoustical systems where a particular electrical energy is now is ready to transform it is already amplified and it is ready to transform to sound in a come back to the sound energy again in and it is now like to propagate and it is called a driver they this calls so loudspeakers or called final driver of the system.

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Use of Loudspeaker

- Aural Communication :**
Used for communicating sound to large audience
- Sound Reinforcement :**
Produce sufficient loud sound to all parts of the auditorium
- Sound Production :**
Various live performance, it supplement some part of musical playbacks
- Sound Reproduction :**
Recorded sound reproduced for some identified events

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So, the what are the use? There are four typical use, it is aural communication we use the loudspeaker. We use loudspeaker for the sound reinforcement, we use sounds the loudspeaker for sound production, for some kind of a live performance and some's many a times we use this is for some kind of a reproduction where a already recorded sound is reproduce by the loudspeaker.

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Type of Loudspeaker

Frequency Range	Shape
Woofer	Horn Loudspeaker
Mid-range Speaker	Cabinet Loudspeaker
Tweeter	Column Loudspeaker
Full-range driver	

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What are the types of the loudspeaker? Again, I have classifier into two categories from the frequency point of view it is having the sorry woofer, the mid-range speaker, the

tweeter and the full-range device and from the shape point of view it is can be classified as the horn, is cabinet loudspeaker or maybe the column loud loudspeaker. So, let us go one by one.

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The slide is titled "Type of Loudspeaker" in a purple box at the top right. It is divided into two main sections. The first section, "Woofers", describes a woofer as a loudspeaker unit for low-frequency bass sounds, covering a frequency range of 20 to 200 Hz, and lists subtypes: Pre-woofer, Woofer, and Subwoofer. The second section, "Mid-range Speaker", describes a speaker for middle level frequency zones, covering a frequency range of 250 to 2000 Hz, and notes it is also known as a squawker. The slide footer includes the IIT Kharagpur logo and NPTEL Online Certification Courses branding. A small video inset of a speaker is visible in the bottom right corner.

Woofers

A **Woofer** is a loudspeaker unit designed to produce extremely **low-frequency bass sounds**.
It mainly covers the frequency range from **20 to 200 Hz**.
Woofers are also further classified as Pre-woofer, Woofer and Subwoofer

Mid-range Speaker

A **Mid-range speaker** reproduces sound in the **middle level frequency zones**.
It generally covers the frequency range from **250 to 2000 Hz**.
It is also known as a squawker.

The woofer is a kind of a loudspeaker which actually very delivered the low frequency and bass sound which is approximately 20 to 200 hertz and they are further classified as the pre-woofer or maybe the subwoofer like that. And, the mid range speakers has them suggest is taken care of the mid frequency level which is 250 or 250 to 2000 hertz and it is also take care of those particular mid frequencies and the delivery properly , but this may not be very much effective for the low or high frequency is also called squawker.

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Tweeter

A **Tweeter** is designed to produce **high frequency sound**.
It typically covers the frequency range from **2000 to 20,000 Hz**.

Full-range driver

A **full-range driver** is a combination of array of speakers designed to reproduce the sound covering and the **entire audio frequency range**.
Full-range driver is mostly adopted in column loudspeaker.

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And, the final one is the tweeter which is the actually design the high frequency sound for design for the high frequency sound which goes with the 200 to 2000 kilohertz sound level also and full-range is for the audio full-range from all the sound level from the mid to high and also the low frequency sound.

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Woofer

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So, this names if you go back if I go back to the names the barking of dog is called woofer. So, it is ah, that is why this low frequency, barking of dog that is why it is called woofer. The mid frequencies are squawkers which is the squawk you know that that is

sound is kind of a mid frequency sound and tweeter that this birds are tweetings and all these are the high frequency of the sound that is why they are called the tweeter kind of the loudspeaker.

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So, this is the from the shape point of view these are the horn loudspeaker which is very popular for the announcement are those kind of a thing. This is the cabinet loudspeaker small and miniature versions cabinet and there are 2 – 3 type of loudspeaker can be placed together and there may be a column loudspeaker where there is a series of low or high frequency or a mix kind of a things also may be placed in a series in a vertical series.

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Loudspeaker Sensitivity

It defines the loudspeaker's ability to effectively convert the electrical power to sound.

Traditionally speaker's sensitivity measures in a standard of 1 watt / 1 meter.

Loudspeaker Sensitivity = 85 dB

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Now, similarly like we deal with the microphones the loudspeaker sensitivities also one of the important things, where it can be defined as 1 watt 1 meter kind of a seen are 1 watt 1 meter kind of a thing; which actually from 1 meter if this power received from in this microphone is 1 watt 1 meter from the loudspeaker and suppose this is producing 85 dB sound then I can see this loudspeaker sensitivity just simple 85 dB.

So, how much I am receiving that should be now fixed, that is, 1 watt and to receive one watt what is the dB level of the this the loudspeaker at a distance 1 meter.

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Directivity

In a loudspeaker system, the **directivity** is an indicator of how **effective** the speaker is at taking the sound it produces and sending it in one **particular direction**.

A loudspeaker that is a **high directivity** device is commonly called a "**long throw**" device.

A speaker with **low directivity** is a "**short throw**" device.

A "**short throw**" speaker system is used to cover the areas **nearer to the loudspeaker**

A "**long throw**" speaker system is used to cover the areas **furthest away from the loudspeaker**.

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And, the next one is the directivity of the sound or the directivity of the loudspeaker, where loudspeaker actually provide kind of a directive sounds in a particular direction. And, from the directivity point of view it is a throw, sometime it is a long throw or sometimes it is a short kind of a throw. So, as you know that the long throws or go nearer to the loudspeaker and the short throw goes to the nearer to the loudspeaker and the long throw was very further away.

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Directivity

The wave length of the sound varies within a wide range, usually 1cm to 18m.

So, the similar directivity or throw for all frequencies is not possible.

Bass frequencies (low frequency) have very **long wavelengths** and it is having **short throw**

The **High frequency** sound have **Short wavelength** and **long throw**

50 Hz

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So, from this point of view it has been seen that it is depend upon the wavelength of the sound and the wavelength of the sound when it is a bass are the low frequency sound which is having a long wavelength it has a short throw something.

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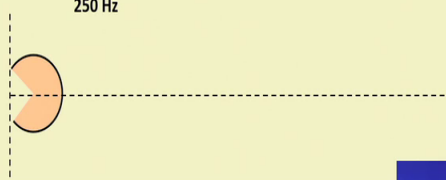
Directivity

The wave length of the sound varies within a wide range, usually 1cm to 18m.
So, the similar directivity or throw for all frequencies is not possible.

Bass frequencies (low frequency) have very **long wavelengths** and it is having **short throw**


The **High frequency** sound have **Short wavelength** and **long throw**

250 Hz



The diagram shows a speaker represented by a semi-circle on the left. A dashed horizontal line extends to the right from the center of the speaker. A dashed vertical line is drawn from the top of the speaker's center to the horizontal line. The sound field is depicted as a wide, semi-circular area extending to the right, indicating a short throw.

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Like this 50 hertz as a very short kind of a throw and then again if I increase it little bit little bit x I mean little bit higher throw, but it is the width narrower width narrow width and the high frequency if you go to the high frequency sound it will be give you a more or shorter a wavelength and it will give a more throw.

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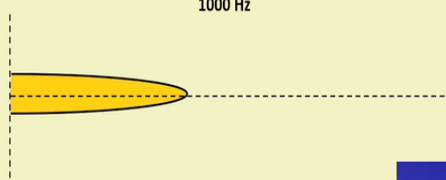
Directivity

The wave length of the sound varies within a wide range, usually 1cm to 18m.
So, the similar directivity or throw for all frequencies is not possible.

Bass frequencies (low frequency) have very **long wavelengths** and it is having **short throw**


The **High frequency** sound have **Short wavelength** and **long throw**

1000 Hz



The diagram shows a speaker represented by a semi-circle on the left. A dashed horizontal line extends to the right from the center of the speaker. A dashed vertical line is drawn from the top of the speaker's center to the horizontal line. The sound field is depicted as a narrow, elongated, yellow shape extending to the right, indicating a long throw.

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Suppose, if you go to the 500 or 1000 it will be narrower it will be narrower, but the throw length will be very high.

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Dimensionless Number = $\frac{\text{Speaker Circumference}}{\text{Wavelength}} = \frac{\pi D}{\lambda}$

$c = 330 \text{ m/s}$

Directivity

Wavelength (50Hz) = 660cm

10 cm

50 Hz

Dimensionless Number = $\frac{\pi D}{\lambda} = \frac{3.14 \times 10}{660} = 0.05$

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So, something like this. So, these also has to taken into some mathematical model cal kind of a calculation with by merge of some kind of a dimensionless number. The speaker circumference by the wavelength has are they have computed has a dimensionless number and they have seen that suppose, if I take the velocity of sound is 330 meter per second and the wavelength for 50 hertz is 660 centimeter and if you 10 centimeter the aperture or the dia of the this particular speaker, so, if I compute the this dimensional number pi D by lambda is 0.05.

(Refer Slide Time: 29:10)

Dimensionless Number = $\frac{\text{Speaker Circumference}}{\text{Wavelength}} = \frac{\pi D}{\lambda}$

$c = 330 \text{ m/s}$

Directivity

Wavelength (50Hz) = 660cm

Wavelength (1000Hz) = 33cm

10 cm

50 Hz

1000 Hz

Dimensionless Number = $\frac{\pi D}{\lambda} = \frac{3.14 \times 10}{660} = 0.05$

Dimensionless Number = $\frac{\pi D}{\lambda} = \frac{3.14 \times 10}{33} = 0.95$

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But, for high frequency sound it is 1000 hertz, the 33 centimeter. So, this particular becomes this particular dimensional number become 0.95 shows it has a higher amount of throw.

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The slide illustrates the concept of a dimensionless number for speaker arrays. It features a diagram of three speakers arranged vertically, with a total height of 30 cm. The sound field is shown as a purple bulb that becomes more directional as the frequency increases. The slide includes the following text and equations:

Dimensionless Number = $\frac{\text{Speaker Circumference}}{\text{Wavelength}} = \frac{\pi D}{\lambda}$

$c = 330 \text{ m/s}$ **Directivity**

Wavelength (50Hz) = 660cm

50 Hz

30 cm

Dimensionless Number = $\frac{\pi D}{\lambda} = \frac{3.14 \times 10}{660} = 0.05$

Dimensionless Number = $\frac{\pi D}{\lambda} = \frac{3.14 \times 30}{660} = 0.14$

The slide footer includes the IIT KHARAGPUR logo and the text "NPTEL ONLINE CERTIFICATION COURSES". A small video inset of the presenter is visible in the bottom right corner.

So, now if I want to go with the 50 hertz you see which is having a low amount of throw and I if I increase the number of speaker in a the horizontal or may be a vertical distance. So, I use three such 10 plus 10 plus 10, 30 centimeter. So, for each individual it has a dimensional dimensionless number is 0.05. Now, if I use three then the dia is increase by 30 and then if you recompute with a same wavelength now it is become 0.14. So, this the purple colour bulb will be the actual throw or the actual directivity of the three, this three.

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Directivity Factor

I_θ = Intensity of Sound at the specific point of interest at radius r_1

By definition, Directivity Factor (Q) is represented by:

$$Q_\theta = \frac{I_\theta}{I_{avg}|_{r_1}}$$

I_{avg} = Average Intensity of Sound over a spherical radius r_1

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So, the directivity is again computed by sound factor called directivity factor. So, from the speaker in a very straightforward 0 degree line, suppose at this particular yellow point I got a the sound level as L average with a sound intensity of I average, but if I go some theta angle with a same radius r_1 this is red I got the red point and I got the intensity as I_θ and L_θ then by definition the directivity factor Q_θ means at particular angle theta is called I_θ by I average at a particular radius r_1 .

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Directivity Factor

As, Directivity Factor is $Q_\theta = \frac{I_\theta}{I_{avg}|_{r_1}}$

Expanding and replacing I_θ & I_{avg}

$$Q_\theta = \frac{I_\theta}{I_{avg}} = \frac{10^{\left(\frac{L_\theta}{10}\right)}}{10^{\left(\frac{L_{avg}}{10}\right)}} = 10^{\left(\frac{L_\theta - L_{avg}}{10}\right)}$$

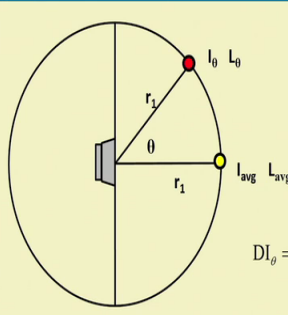
Directivity Factor expressed in terms of SIL $Q_\theta = 10^{\left(\frac{L_\theta - L_{avg}}{10}\right)}$

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So, is particular computation if you can do, now the I_θ can be rewritten as this I format and the I can be written as that this way the I reference into 10 into 10 to the into 10 to the power L_θ by 10. So if I use this I_θ and I_{avg} and if I replace this by this level sound pressure level or the sound intensity level I give this as a final result. So, the in terms of the SIL sound intensity level Q_θ can be rewritten as in this yellow box format so, 10 to the power if this particular the L_θ minus L_{avg} by 10.

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Directivity Index




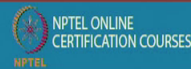

By definition, Directivity Index (DI) is represented by:

$DI_\theta = 10 \log(Q_\theta)$

$$DI_\theta = 10 \log(Q_\theta) = 10 \log \left[10^{\left(\frac{I_\theta - I_{avg}}{10} \right)} \right] = 10 \times \left(\frac{I_\theta - I_{avg}}{10} \right) \log 10 = (I_\theta - I_{avg})$$

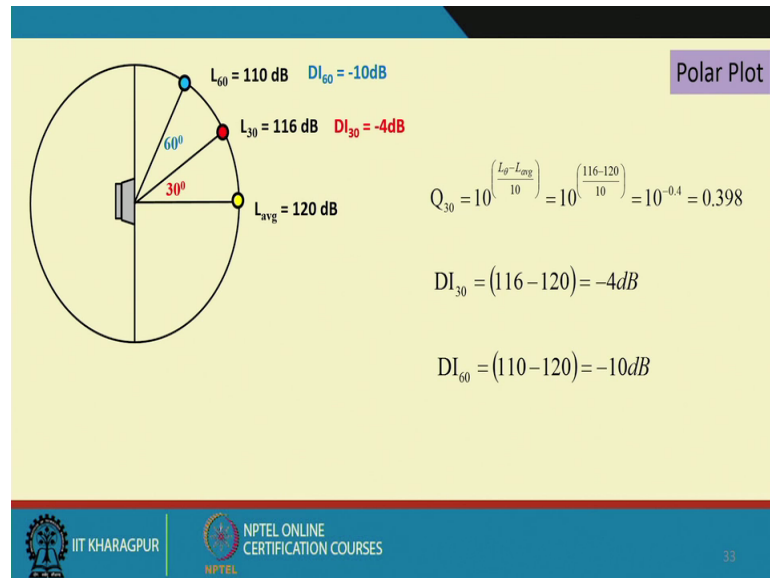
Directivity Index expressed in terms of SIL

$DI_\theta = (I_\theta - I_{avg})$

So, next is the directivity index directivity index is DI of that particular theta is nothing, but the 10 times logarithmic of this directivity factor Q_θ and if I re open it this particular equation it is nothing, but the L_θ minus L_{avg} . So, whatever may be the level of here and whatever may be the L_{avg} of here the difference between them is the directivity index.

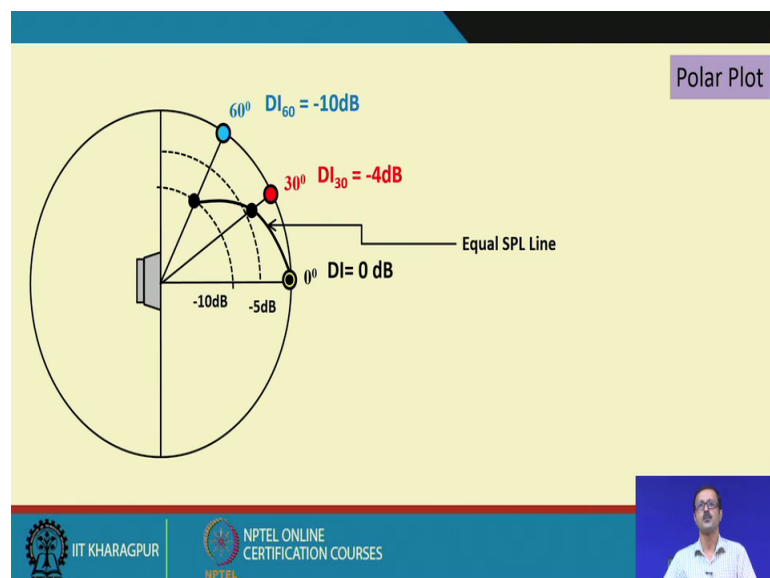
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So, let us go to a small mathematical understanding where suppose in this particular 0 degree line I got a 120 decibel is the L average and if I move circular in a circular path without change in the radius from the loudspeaker at 30 degree I got 116. So, you are directivity index is a minus 4 dB because 116 minus 120 is minus 4. Similarly, in L 60 degree it is suppose 110, so, it is minus 10 dB.

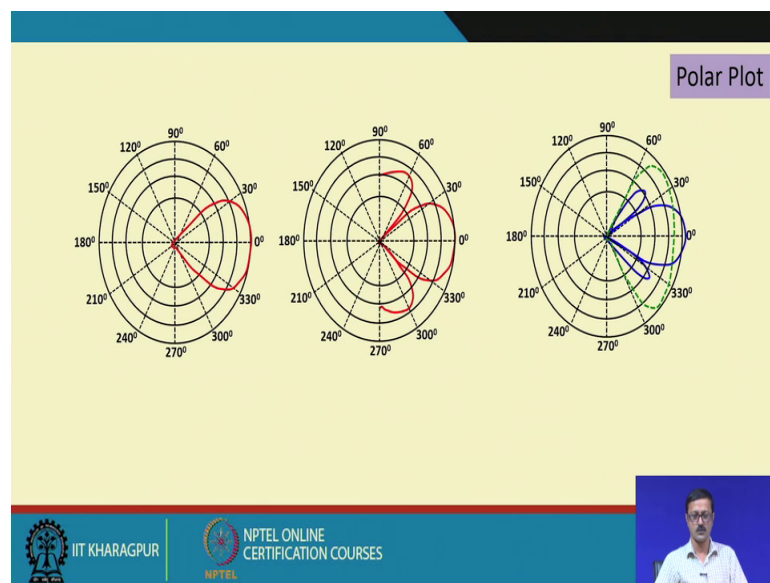
So, what it can give me is that in the same radius if I move gradually the decibel level is slower down.

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And, if I now go with this, so, at particular 30 degree it has minus 4 and this is minus 10 so, I can say a I can draw a concentric circle of minus 5, minus 10 dB something like that and then I can say this equivalent sound level of this is nearly somewhere here and equivalent sound level of this yellow which is in the 0 degree line is nearly here in the sixty degree line because it is minus 10; minus 10 from the this particular and this is minus 4 and finally, if I draw this in a curve I mean draw this particular points in various circular segments in curve I can I can actually draw equal SPL line and this is called the polar plot of the polar plot of the loudspeaker.

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So, there are this kind of polar plot can be drawn. So, in the left side this is the this is a single polar plot are this particular microphone I am sorry the loudspeaker has a this kind of a directivity and this is the equal SPL contour this is a maybe two or three microphones or may be a single microphone has this kind of a direct little bit of complex directivity, where these are the polar plot of the watt particular loudspeaker. And, this loudspeaker has I have drawn blue and green. Suppose, this blue and green or for may be two different frequency suppose this is some high frequency and this is suppose some kind of a low frequency kind of a directivity and the equal SPL plots.

So, this equal SPL plots comes from the directivity which is a one of the significant property or the parameter of the loudspeaker which we will again take into the next chapter next discussion lecture number – 27 for further details.

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Home Work

Differentiate between Microphone and Loudspeaker sensitivity

How the Directivity and Polar Plots of Loudspeakers are related. Also state its importance in electro-acoustical design.

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End of Lecture 26: Electro-Acoustics - I

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The slide features a yellow background with a blue header and footer. A purple box in the top right corner contains the text 'Home Work'. Two white boxes with black borders contain the homework questions. The footer includes the IIT Khargapur and NPTEL logos, and a small video inset of a man in a white shirt.

So, this lecture ends here. So, let us take some the homework and this two homework I have design it is not a mathematical it is just kind of a discussion or this kind of a understanding. So, can you differentiate between the microphone and the loudspeaker sensitivity and these are little that is a little difference between them the loudspeaker sensitivity and the microphone sensitivity. And, you can another one is the how this directivity of the polar plots or the polar plots of a loudspeaker related, directivity and the this polar plots how they are related and what are the importance of this polar plots do you think for the electro acoustical design.

(Refer Slide Time: 35:45)

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End of Lecture 26: Electro-Acoustics - I

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The slide features a yellow background with a blue header and footer. A purple box in the top right corner contains the text 'Bibliography'. A list of four references is presented in black text. The footer includes the IIT Khargapur and NPTEL logos, and a small video inset of a man in a white shirt.

So, that is ends my lecture these are the sum of the books I have referred for this lecture and next lecture in that lecture number 27, will go with the some acoustical design principles.

So, thank you for joining us and let us move to the next lecture.