Digital Speech Processing
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Lecture - 23
Autocorrelation Method Of LPC Analysis (Contd.)

So last class; we are here that alpha is equal to R inverse; small r.

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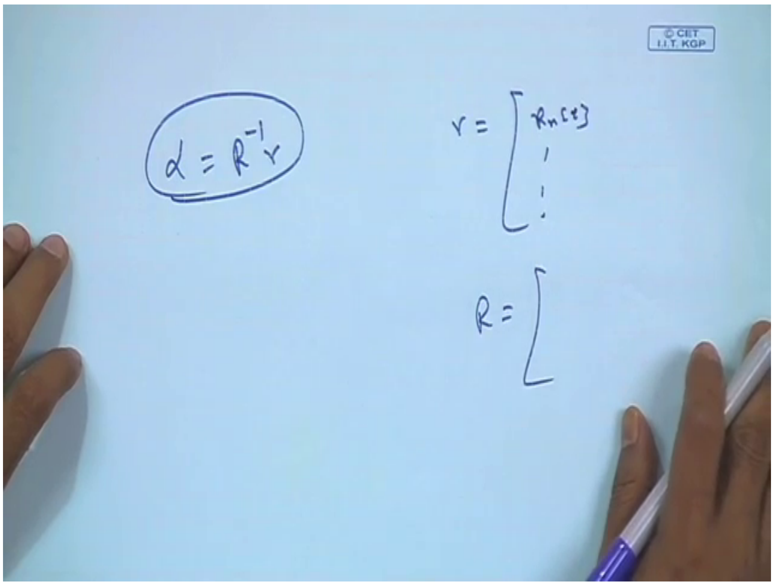

So, I can say the small r; the small R is nothing but a small R is equal to R n 1, R n 2 like that and R is nothing but a this matrix. So, R is this matrix and small R n is this matrix; so, this matrix and this matrix. Now, we have to solve this; how to solve this? Solving is done based on the Levinson and Durbin methods. So, what is this method? Basically I am not details describe the method, but let us go through that.

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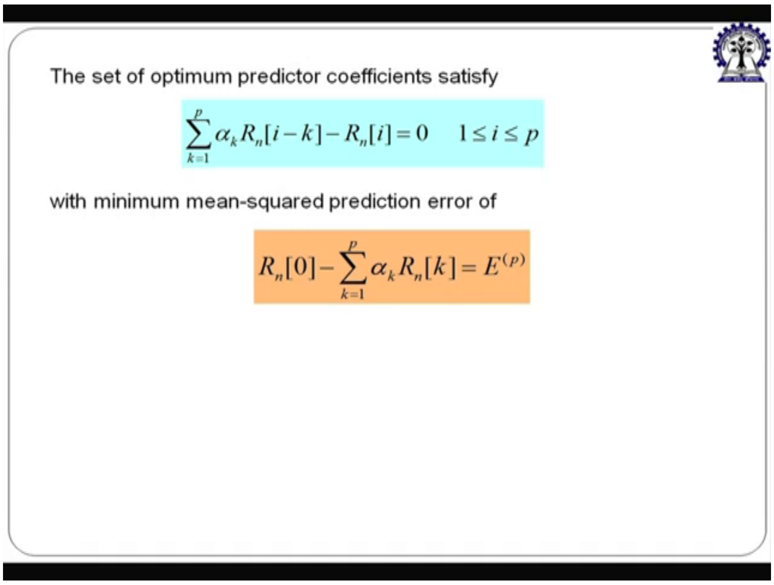

So, basically these equation can be solved to set up optimize predictor coefficient; satisfy if I solve this basically I am optimize the alpha value will satisfy these two equation.

So, these two equation derived; so, if I write down that these two equation in matrix form; the first equation is give this matrix and second equation if I included the minimum prediction error; that means, if I estimation is very correct, then it should produce the minimum prediction error. So, minimum prediction error equation it should satisfy. So, if I put this two equation in a matrix form it will look like this.

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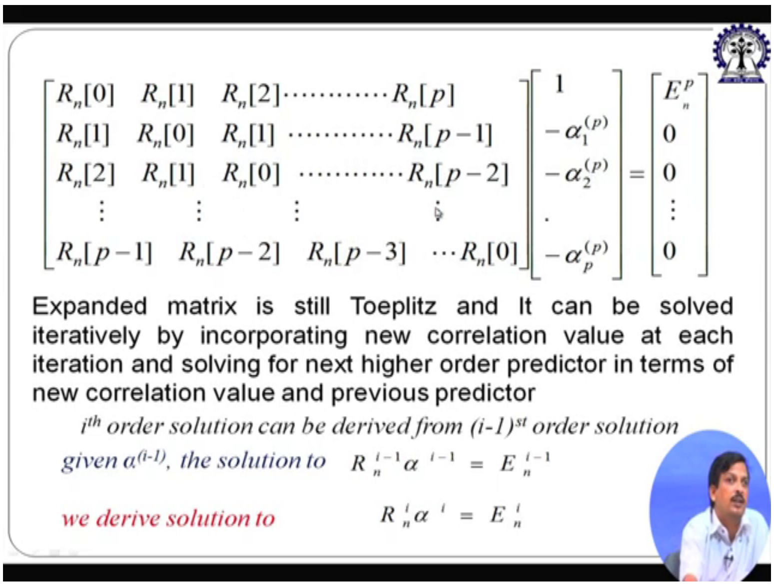

1 enp; because some matrix that this equation to many matrix form. Now, if I see I want to solve this matrix; it shows that if I use the Levinson recursion; that means, that means the ith order solution can be derived from i minus first order solution. So, suppose I have a P is my level of prediction; P is the number of predition that I want the order of the predictor is p. So, I can say I can iteratively predict the alpha value and I want that if I iteratively predict it; then at pth order will give me the optimum solution or pth iteration give me the optimum solution.

So, in Levinson recursion I said we can say that I can predict the current i value or current this matrix equation; current value from the previous or ith order solution. So, what is the i minus 1th order solution?

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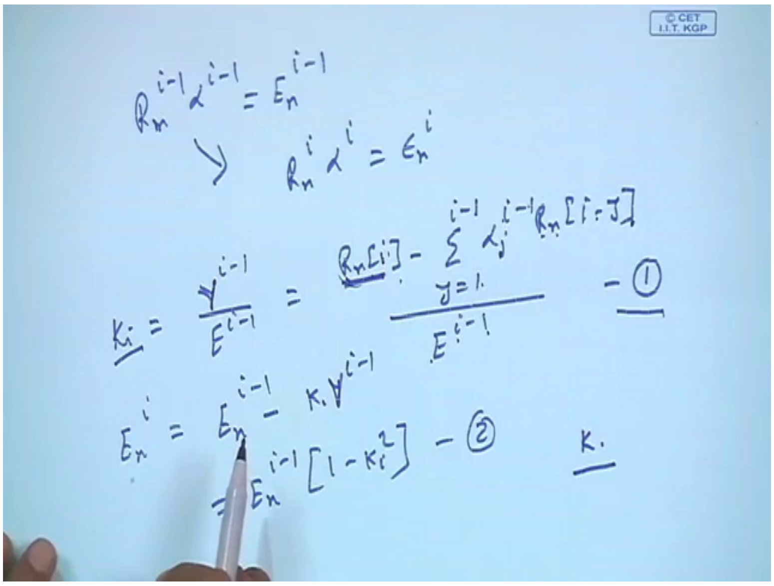

R n; i minus 1 alpha i minus 1; is equal to E n i minus 1 if I see this matrix. So, from here we can derive the solution for R n;i alpha i is equal to E n i this we can derive.

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The $(i-1)^{st}$ *solution can be expressed as:*

$$\begin{bmatrix} R_n[0] & R_n[1] & R_n[2] \cdots\cdots & R_n[i-1] \\ R_n[1] & R_n[0] & R_n[1] \cdots\cdots & R_n[i-2] \\ R_n[2] & R_n[1] & R_n[0] \cdots\cdots & R_n[i-3] \\ \vdots & \vdots & \vdots & \vdots \\ R_n[i-1] & R_n[i-2] & R_n[i-3] & \cdots R_n[0] \end{bmatrix} \begin{bmatrix} 1 \\ -\alpha_1^{(i-1)} \\ -\alpha_2^{(i-1)} \\ . \\ -\alpha_{i-1}^{(i-1)} \end{bmatrix} = \begin{bmatrix} E_n^{i-1} \\ 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

How is derived? That is the details explanation that matrix I can write this way and then I can append 0 vector at alpha multi; this is the matrix, you can see that this ride and you can find out this is the nothing but a mathematics; nothing gains in the mathematics. So, I can say gamma minus i to the power one is equal to this and then I can say this is the toeplitz matrix; special symmetry, we can reverse the order of the equation we can reverse it and then once you revert you can combined into set of multiplication factor R n this is minus and this is minus both side.

So, it remains same and then we choose i minus 1. So, that the vector this K i right side has a only a single non zero entry. So, K i it is the wide vector of the predictor is nothing but a gamma i minus 1; divided by E i minus 1 where it is nothing but a R n i minus j equal to 1 to i minus 1 alpha j; i minus 1 R n; i minus j, divided by E of i minus 1; this is the equation number 1 for autocorrelation method solution ki.

Now, if it is this is ki; then what is the E n i is nothing but a; if I see E n i, I can write down E n i is nothing but a R 0 minus alpha K R n K ik. So, if I use that value then E n i becomes E n i minus 1 minus K i gamma i minus 1; which is nothing but a E n i minus 1; 1 minus K i square.

So, ith K i is nothing but this R n i minus. So, R ni means R n is the autocorrelation coefficient; ith autocorrelation coefficient minus j into 1 to i minus 1, alpha j i minus 1 R n i minus j divided by E i minus 1; this is the equation number 1, this is the equation number 2; then I can put that value and this K i is called partial correlation coefficient.

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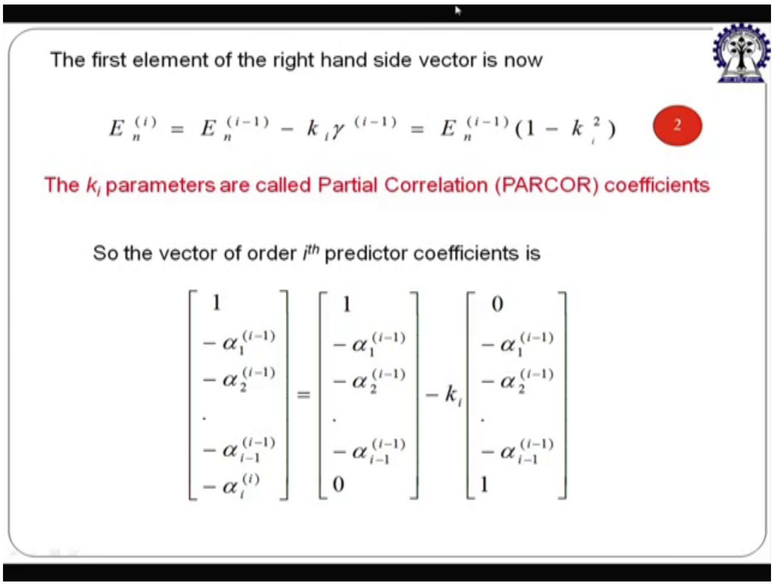

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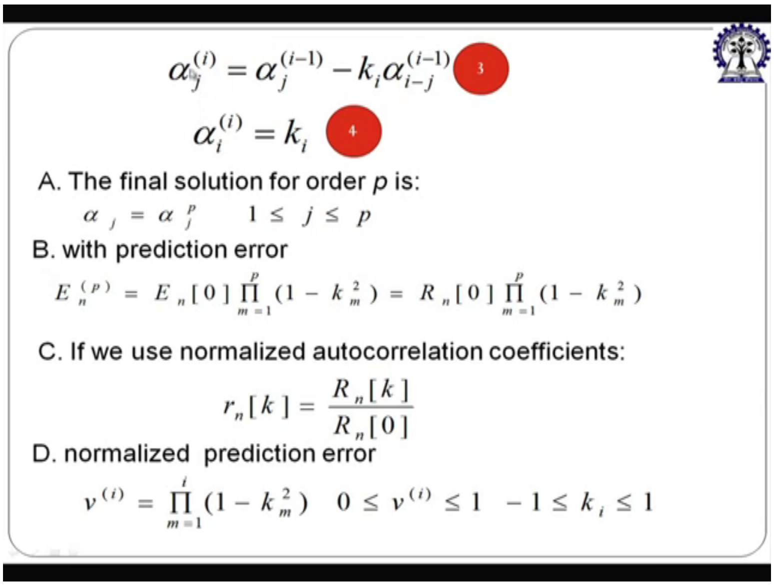

Then we can solve for alpha ji and this alpha ii K i this is the forth equation. So, final solution for a order P if my LPC order is p.

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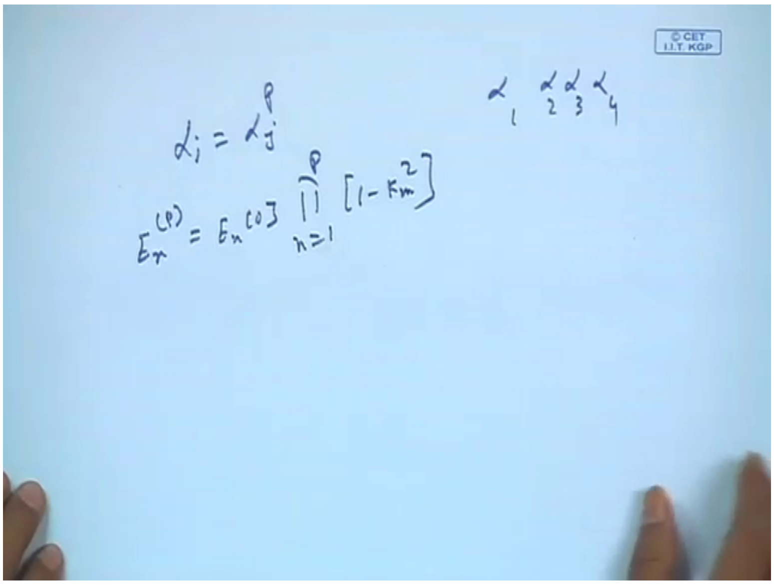

So, alpha j is equal to alpha j P or I can say k; my alpha j P is the my last set of alpha value or optimal set of alpha value alpha 1, alpha 2, alpha 3, alpha 4; optimal set of alpha value with a prediction error E n P is equal to E 0, E n 0 energy n equal to 1 to P 1 minus km square.

So if I write down this four equation implement this four equation in a softwares; then I can calculate that alpha value and I can calculate the prediction error. So, if you see the next slides; so, there is a equation.

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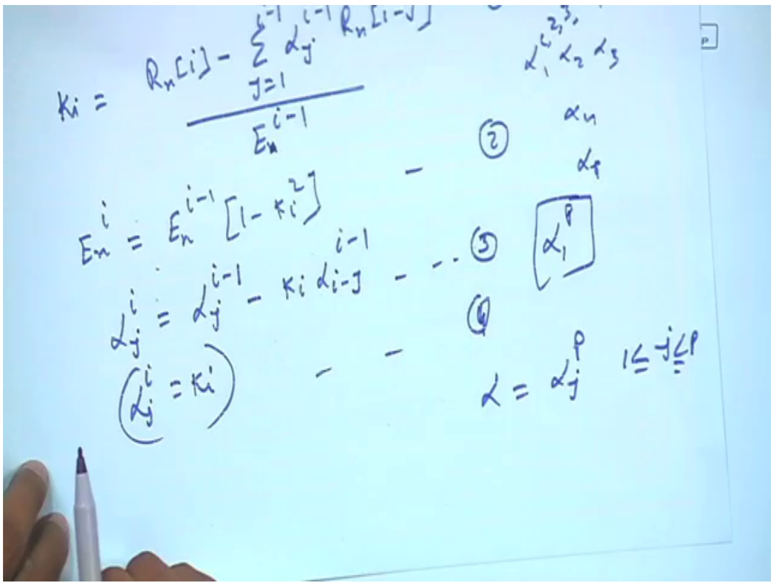

So, if I say here; if I write down the four equation one is K i is equal to; if I write down the equation K i is equal to R ni minus j equal to 1 to i minus 1; alpha j i minus 1, R n i minus j divided by E n or E i minus 1; what is E n i? E n i is nothing but a E n i minus 1 into 1 minus K i square. So, this is equation number 1; this is equation number 2; then I can say alpha ji is equal to alpha; j i minus 1; minus K i alpha i minus j; i minus 1 and alpha ji is equal to ki; this is the 3, this is the 4 equation.

So, now suppose I have a speech segment and I want to find out this value; K i E n i and ultimately I have to find out this value. So, at the end alpha value is nothing but a alpha j P; the P is the order of the alpha j P; where j is also 1 equal to j P; j varies from 1 to P. So, what I want? I want a set of alpha 1, alpha 2, alpha 3, alpha 4, alpha p, but those set of alpha value can be extracted in iterative method; alpha 1, 1; alpha 1, 2; alpha 1, 3 to alpha 1 P. So, alpha 1 P is my final answer; so, using this four equation this I can do. So, how I can do it? I just take up that let us take a simple example.

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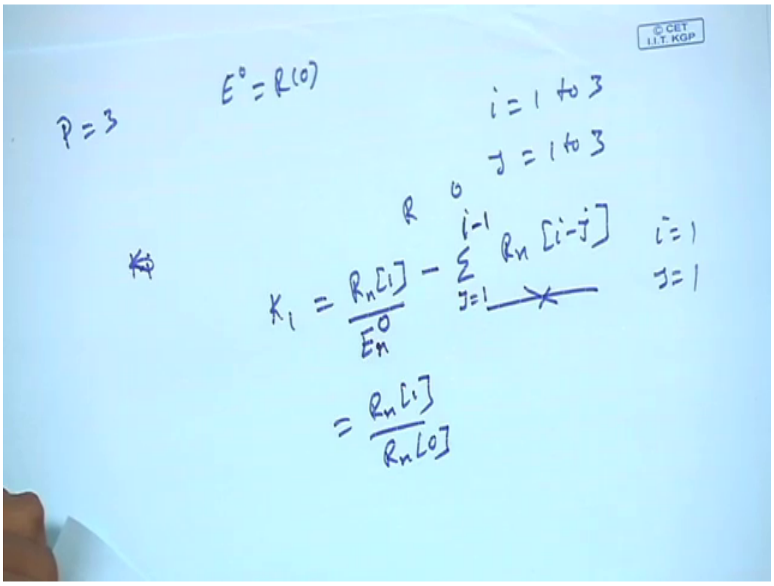

Let us; I want to find out the order of the predictor P is equal to 3; so, i is varies from 1 to 3 and j is varies from 1 to 3. Now, if I see ki; so, what is R i is; sorry i is varies from 1 to P and j is varies from 1 to p. So, if I see; I want to find out K 1; if I use this equation I want to find out K 1. So, what is k? K 1 is equal to R n 1; R n 1 minus j is equal to 1 to i minus 1; R n i minus j.

Now, here I say I is equal to 1; so, j is equal to 1. So, if it is j is equal to 1; then i 1 minus 1 is 0. So, this will not is equated that; so, this R 1 divided by E n i minus 1; E 0 j minus 1 is 0; E 0. So, what is E 0? This is the energy which is nothing but a R 0. So, I can say R n 1 divided by R n 0; now I can find out K 2; K 1; I find out, then what is E n i; 1 is equal to E n 0 into 1 minus K 1 square.

So, I get the value of K 1 here; I can put the value K 1 here and find out E 1 n; then what is alpha j i? If you see alpha ji here I put; i equal to 1. So, it is alpha 1; 0 minus K 1, alpha 0 0; alpha 0 0 is equal to alpha ji 1, 1; alpha 1 0; alpha minus K 1; 0 0, I can get the K alpha 1 and alpha 1, 1; then I can say alpha 1, 1 equal to K 1. So, alpha 1, 1 is equal to K 1; so, this way; this is implemented in here.

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$$\mathcal{E}^{(0)} = R[0]$$

for $i = 1, 2, \ldots, p$

$$k_i = \left(R[i] - \sum_{j=1}^{i-1} \alpha_j^{(i-1)} R[i-j] \right) / \mathcal{E}^{(i-1)}$$

$$\alpha_i^{(i)} = k_i$$

if $i > 1$ then for $j = 1, 2, \ldots, i-1$

$$\alpha_j^{(i)} = \alpha_j^{(i-1)} - k_i \alpha_{i-j}^{(i-1)}$$

end

$$\mathcal{E}^{(i)} = (1 - k_i^2)\mathcal{E}^{(i-1)}$$

end

$$\alpha_j = \alpha_j^{(p)} \quad j = 1, 2, \ldots, p$$

If you see; this is the implementation the program is here. So, you can write the program using this four equation. Now, sometime you may find out that is called normalize autocorrelation; so, normalize, if you see the autocorrelation; the value of reflection coefficient or K i or alpha i; both are depends on R value.

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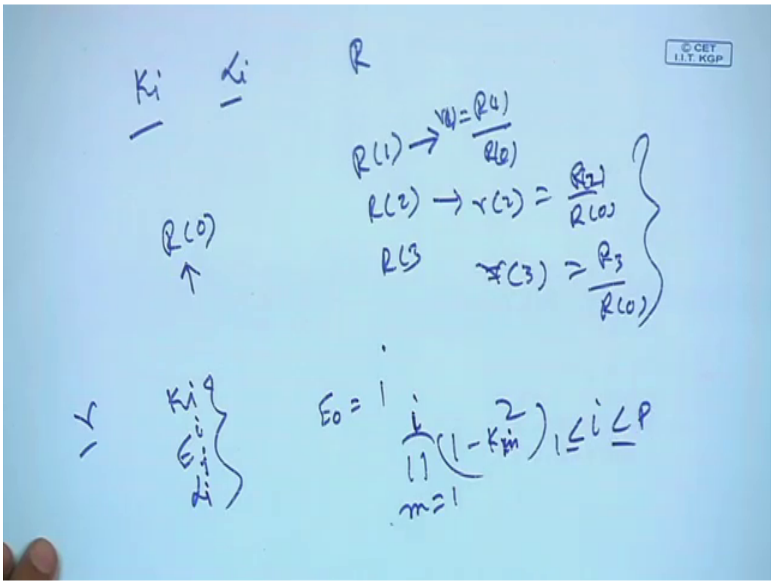

Autocorrelation; value of autocorrelation; so, sometime if you find that value of if it is magnitude its change of the speech signal, then may be R value will be change and autocorrelation coefficient may slightly changed. So, what I want? I want normalized autocorrelation that with respect to R 0; I can normalize that correlation value. So, suppose my correlation value is 1; R 2, R 3.

Now, I can normalize R 1 with respect to R 0; because R 0 is the energy of the signal. So, I can say R instead of R 1, I can use small r 1 which is nothing but R 1 by R 0. R 2 small r 2 is nothing but a R 1 by R 0; R 2 by R 0. Similarly, R 3 small r 3; R 3 by R 0; so those are called normalized then if I use this small R to the extract the value of K i and E i; E i and alpha i j, then I can say this is normalized autocorrelation. So, if I say what is the normalized prediction error? If it is normalized, then prediction error E 0 is equal to 1 because it R 0 y R 0. So, it is nothing but a m equal to 1 to i; 1 minus K i km square I varies from 1 to P.

So this is the normalized autocorrelation; so, this is the program you can implement it in find out whether it is happen or not.

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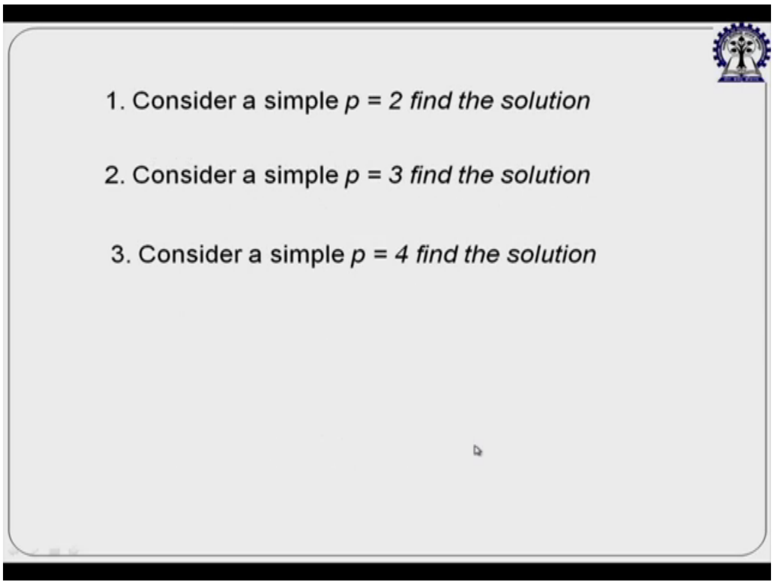

Then that is the task I have given; consider a sample P is equal; second order P is equal to find the solution. So, I can give you let us xn is equal to I give you not speech signal.

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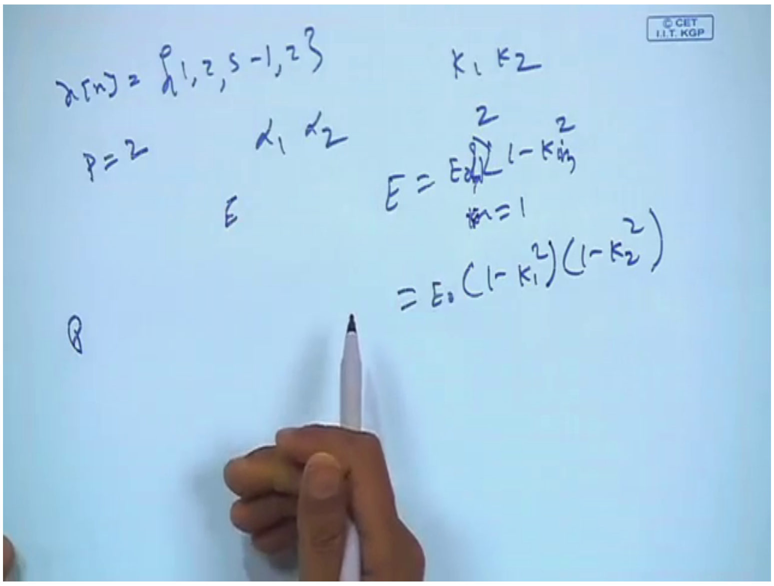

That is 1, 2, 5 minus 1, 2 like that and then I told you that for P is equal to 2; find out that alpha 1 and alpha 2 and find out the autocorrelation error; E.

So, you can find out K 1, K 2; so, I can say E is equal to nothing but E 0 into 1 minus K i square where i varies from; I can say 1 to m square; where m equal to 1 to order 2. So, it is nothing but a E 0 into 1 minus K 1 square into 1 minus K 2 square; and I can find out alpha 1 to alpha 2 using this equation in hand made without programming.

So, if P value are very less without programming I can find out that alpha alpha 2. So, those kind of questions can be you expect in that examination paper or I can write a program and run as real speech signal and find out the that alpha value and E value for autocorrelation function, using the autocorrelation coefficient. So, those are the autocorrelation solution.

Now, how do you define the order of the P? So, I say prediction order. So, how you which factor the P is depended; prediction order I have take a signal which is 8 kilo hertz sampling rate I have take a signal which is 16 kilo hertz sample; I have take a signal which is 32 kilo hertz sample. So, make different sampling rate; I can sample the speech signal.

Now, what kind of order I should use to get that correct coefficient value; if it is true that if I increase the order, your error will be less. So, I cannot take infinite order. So, what kind of order is optimum; so, in what basis I can decide the order?

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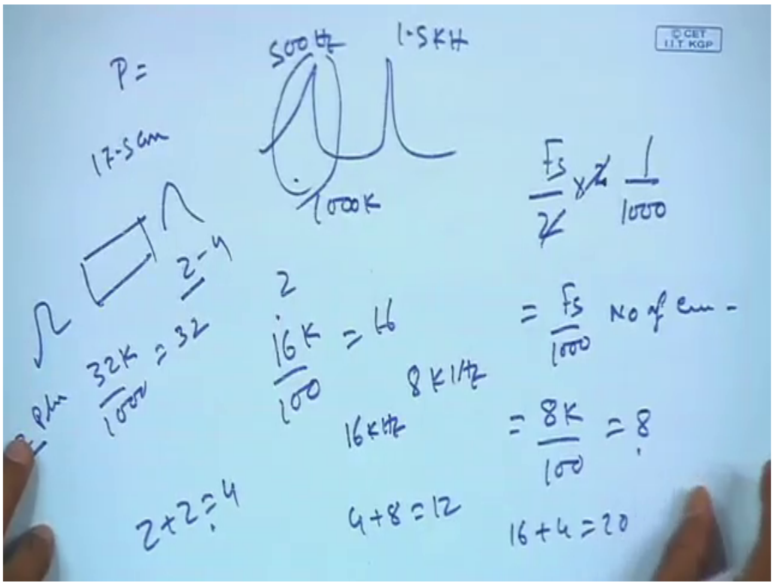

So, if you see that in tube model for formant; every formant are separated by 1 kilo hertz; 1 kilo hertz signal. So, if you see; if you remember in the first fist class; the first tube model uniform tube model; if tube length is 17.5 centimetre, first formant is 500 hertz; next formant is 1.5 kilo hertz.

So, they are separated by 1000 hertz; that means, 1 kilo hertz; so, I can say that if it is my F s is my sampling frequency. So, what is the basement frequency? F s by 2. How many complex pole is required to realize one formant? 2; complex conjugate pole is required to realize one formant; so, it is 2. So, F s by 2 into 2 pole; how many? Every 1000 hertz signal; so, I can say F s by 2 into 2 by 1000; 2; 2 cancel. So, I can say F s by 1000 number of pole; number of complex conjugate pole; complex pole not conjugate, conjugate then it is will be have; so, complex pole.

So, if it is 8 kilo hertz; then 8 K divided by 1000; so, I can say 8; if it is 16 kilo hertz then 16 K divided by 1000; 16. If it is 32 kilo hertz 32 K divided by 1000; so, it is 32. So, those are for formant; that means, those are for tube only. Next is there is a glottis and there is a radiation laws; for glottis 2 to 2 poles and for radiation laws; 2 to 4 poles. So, if I say radiation 2 pole and glottis 2 pole; so, 2 plus 2 is equal to 4 pole; 4 plus 8; 12 pole, if my signal is sampled at 8 kilo hertz.

If my signal is sampled by 16 kilo hertz; then I can say 16 plus 4; 20; 16 plus 20 or 21 or 22 I can say; 16 plus 6 also; if I glottis if the radiation laws is some used by 4 pole. So, order of that LPC analysis; it is not arbitrary, it depends on the sampling frequency. So, based on the sampling frequency; I can take the order and find out the LPC analysis.

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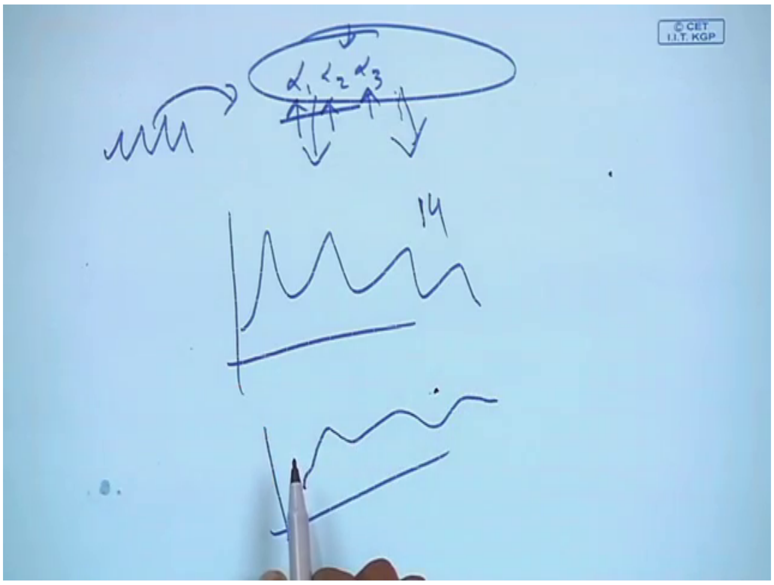

Now, there is another question is there LPC analysis; then if if I have the speech signal; I extract that filter LPC analysis give me the alpha 1, alpha 2, alpha 3 value. Those actually characterize the mobile track; now forget about this part I will come later on then if those coefficients are represent; you can say the pole position.

So, formant those are the represent the actual pole position. So, I can say if I take the frequency task form of those coefficient; I should get that LPC spectrum which will give me the resonant frequency; have you understand or not? So, if I say F s by 2; number of pole I said now those coefficient are represent the pole position of the speech signal. Now if I take the spectrum or the frequency analysis of this LPC coefficient; this would give the LPC spectra; they should represent the peak at forward frequency.

Now, if order of the analysis is reduced then this should be. So, suppose I required let us 14 order I make it 12 order. So, some of the formant will combine together and give me a some broad kind of structure. So, if you see in this picture

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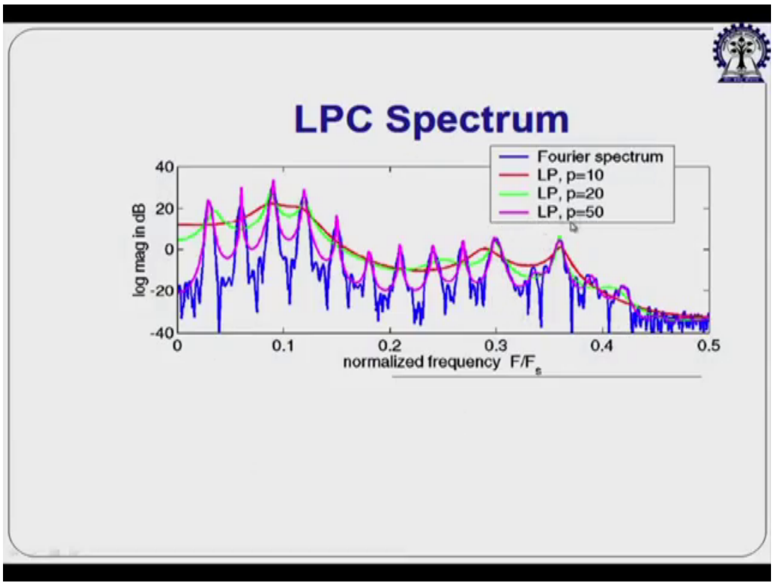

If my LPC order is increased is more or less; copy the actual blue colour is the actual spectrum. And if you see the red colour why the LPC analysis is 10; so, there is a lot of formants are there. So, I can see that; it is roughly estimate that spectral envelope, if I increase the LPC order; the number of variation is increases. So, it accurately increase the; it accurately copy the spectrum envelope. So, depending on my requirement if I want the smooth spectrum; I do not want this lot of variation, then I can use LPC order reduce the LPC order if I want that I want to exactly copy the spectrum, then I can increase the LPC order.

So, using this I can also draw the spectrogram also this is the real life example of LPC spectrum.

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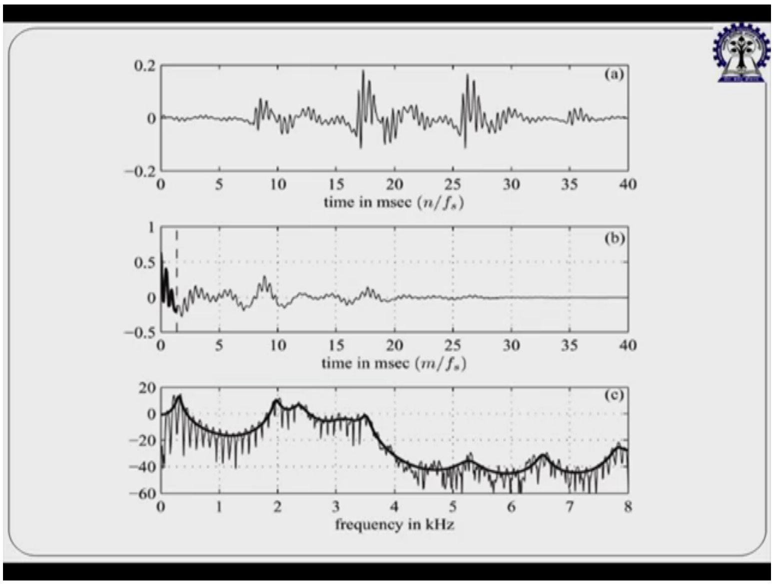

If you see how many formants are there 1, 2, 3, 4, 5, 6, 7; now if I say what is the possible order of the LPC analysis?. So, I can say there is a 7 formant. So, I can say 7 into 2; each formant require two complex pole. So, it is nothing but a 14 order LPC analysis.

So, this can be explained; this is called LPC spectrum and then there is a LPC spectrogram can; this is the frequency analysis equation, you can go through that slides I am not details describe this frequency analysis equation and this is the LPC spectrogram, because here if you see the formants are much more clear; this is done, how it is plotted?

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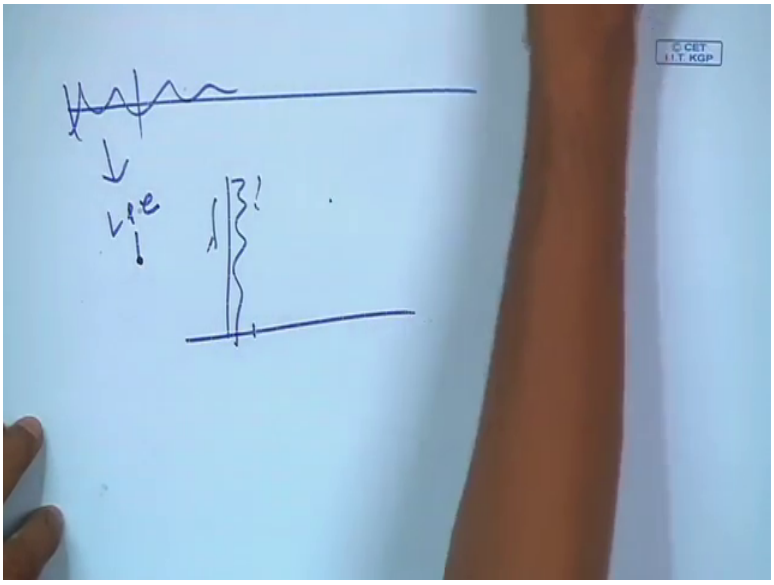

Suppose you have a speech signal; you take a window, find out the LPC coefficient take the spectrogram of the LPC spectrum of the draw the magnitude spectrum of the LPC coefficient. Then for that time plot this in this frequency scale; again shift it and plot it and that way you get the LPC spectrogram.

So, the LPC spectrogram now formant; suppose that this I will discuss later on. Also I have discuss in that the during the tube analysis that formant position R E to the power j theta. So, frequency of the formant is given by F s by 2 pi into theta.

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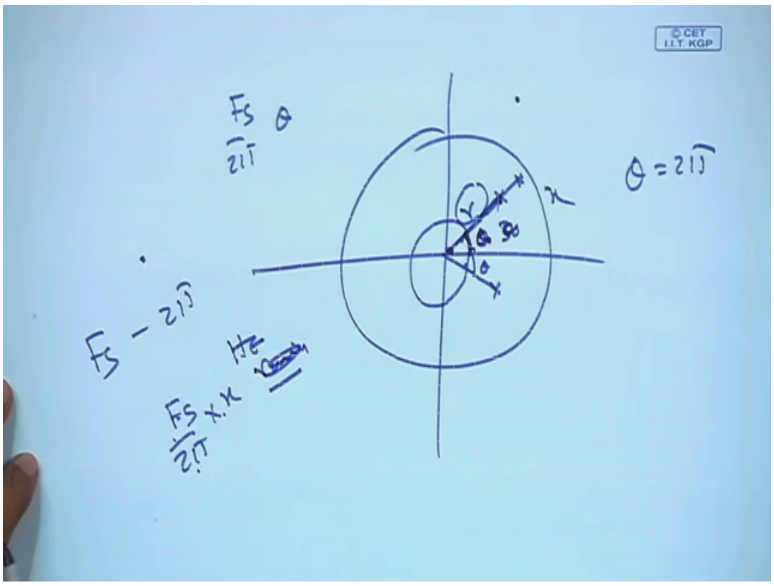

So, if I say every complex conjugate pole indicate one formant; if it is close to unit circle this is called formant frequency it contribute to the formant frequency because energy will be very high.

So, if I say complex conjugate pole**.** So, suppose this is my unit circle and if there is a formant if there is a pole angle is theta. So, the system is real; so there will be a complex conjugate pole minus theta here and this is r. So, r which if it is close to unit circle the bandwidth r will be increase and bandwidth will be increased; if it is close to 0 the bandwidth will be 0. So, if you see the formant frequency is determined by this angle theta and the formant bandwidth it determined by R; which is already explained in tube modelling.

So, if F s is my sampling frequency; so, what is the normalized frequency of the digital signal? Is 2 pi is equal to F s. So, what is the value of the theta; theta maximum value is 2 pi and which is equal to F s. So, if the theta is let us 30 degree or this radian; if it is radian let us say it is x radian then I can say F s by 2 pi into x is the radian is the frequency, if it is hertz then I can convert F s is in hertz. So, I can say F s by 2 pi into x hertz is my formant frequency. Suppose I have a pole I do not know whether the example is here or not; no example is not there.

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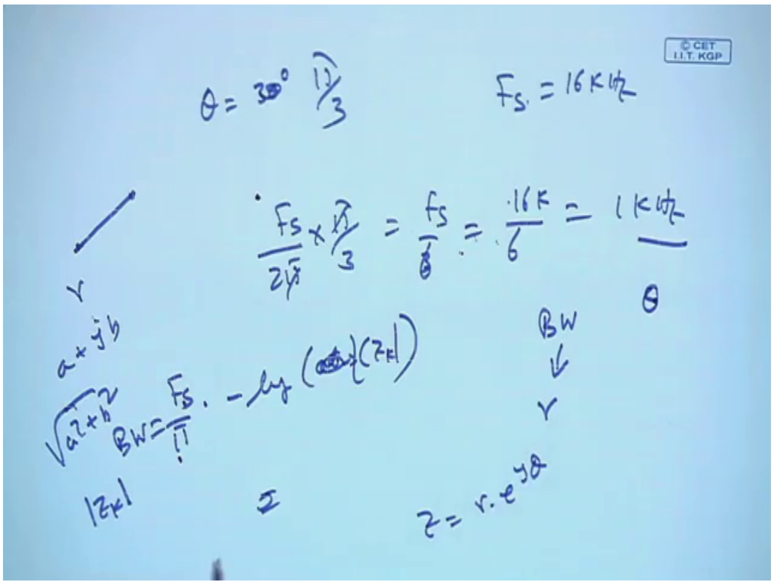

So, suppose the theta is equal to let us 30 degree or let us pi by 3. So, what is the formant frequency? So, let us say F s is equal to 16 kilo hertz. So, if it is 16 kilo hertz then F s by

2 pi into pi by 3. So, F s by 6 is nothing but a 16 K divided by 6; it is 1 kilo hertz by formant frequency and formant bandwidth is nothing but a F s by pi into log of this is Z r log of r; you know that; log of r what is r? r is nothing but a absolute value. So, if it is complex pole a plus jb; then I can say root over of a square plus b square or mod of if it is pole is Z K mod of Z K; give me the r. So, I can say pi by F s by pi into minus log of abs zk or mod of zk.

So, I can find if the r value I know, if I know the sampling frequency if pi I know. So, you can got the I get the formant bandwidth. Or vice versa if I know the formant bandwidth, I can find out the value of r; if I know the formant frequency I can find out the value of theta then I can derive the transfer function of that thing. Because Z is nothing but a r into E to the power j theta or you can use that r theta equation in tube model. So, both way if I know the formant bandwidth and formant frequency; I can derive the transfer function; linear LTI transfer functions. Or if I know that pole position or pole value or theta and r; I can derive the formant frequency and bandwidth.

So, this is autocorrelation method I have described and all other things I have described. Then there is a other methods also that if you see that there is a covariance methods for linear prediction. So, covariance methods is also one another methods instead of autocorrelation, I can find out the covariance method to detect the; to find out the LPC coefficient. So, what you want? We want the same things same solution of that matrix solution of this matrix; this we have already derived.

Now, that difference is that; in this matrix that I have not taken the outside the window signal.

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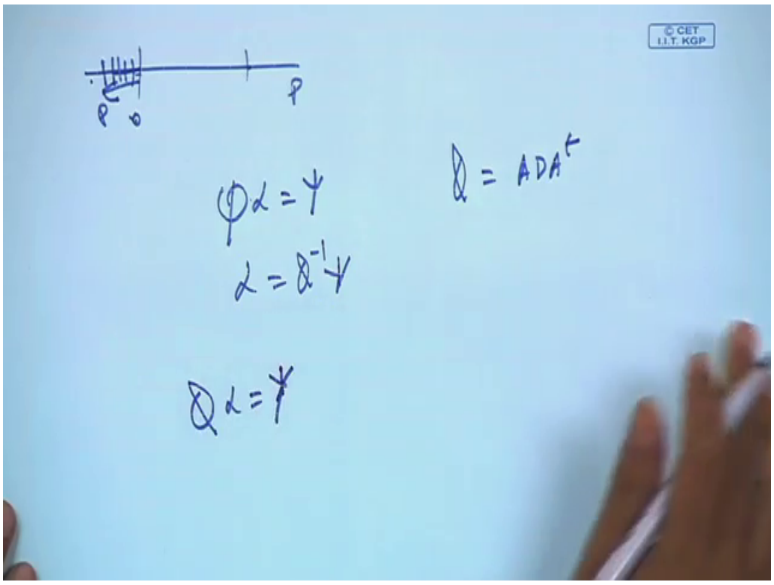

Is 0 if I take it then problem will come; so what we will take? The key difference model method is that limit of summation include the term before m equal to 0. So, if the order is P instead of 0; I will consider the previous sample also and here also I compute previous sample also I have to consider. So, if the order is P; P number of previous sample I have to consider.

So, tapering of the window does not matter here the window does not matter here whatever window function I can use, but I should say that P number of previous sample is required in this analysis. Then same simple things; I have to solve this matrix equation. So, phi you can say this psi into alpha is equal to let us this one; then I can say alpha is equal to phi inverse or I can rate this is not phi; how to solve this?

Now, this can be solved called Cholesky decomposition methods who is that; let phi is equal to matrix A D A transpose and phi into alpha is equal to let us this one. If phi is this matrix; where A is equal to lower triangular matrix and first in the main diagonal and D is the diagonal matrix and A transpose is the upper triangular matrix. So, I can say; so instead of let us solve for P equal to 4; order is equal to 4.

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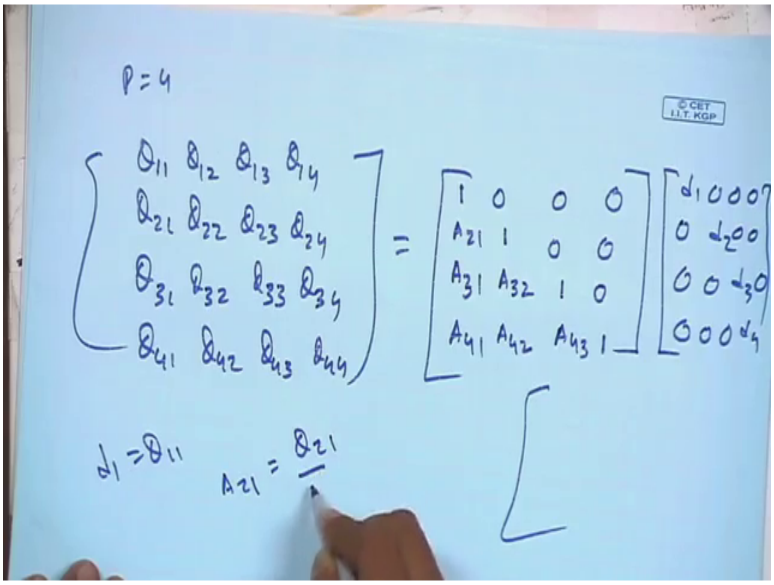

So, I can get phi equation is phi 1, 1; phi 1, 2, phi 1, 3, phi 1, 4; phi 2, 1, phi 2, 2 phi 2, 3; phi 2, 4; phi 3 1, phi 3 2, phi 3 3, phi 3 4 phi; 4 1, phi 4 2, phi 4 3, phi 4, 4. This matrix let us equal to what triangular matrix; that is diagonal element is 1; so, 1 then it is A 2 1, A 3 1, A 4, 1; then I can say put 1; I can say A 3, 2 A 4 2; then I can put one; I can say A put 3; then I put 1 here; rest are 0, 0, 0, 0, 0, 0 this into let us diagonal element d 1, d 2, d 3, d 7; 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0. So, diagonal element and a transpose if it is lower triangular is the transpose is upper triangular matrix.

So, this is this; now if I solve this, then I can get solve this matrix, if I solve this d 1 is nothing but a phi 1, 1. Similarly A 2 1 is nothing but A phi 2 1; divided by d 1; A 3 1.

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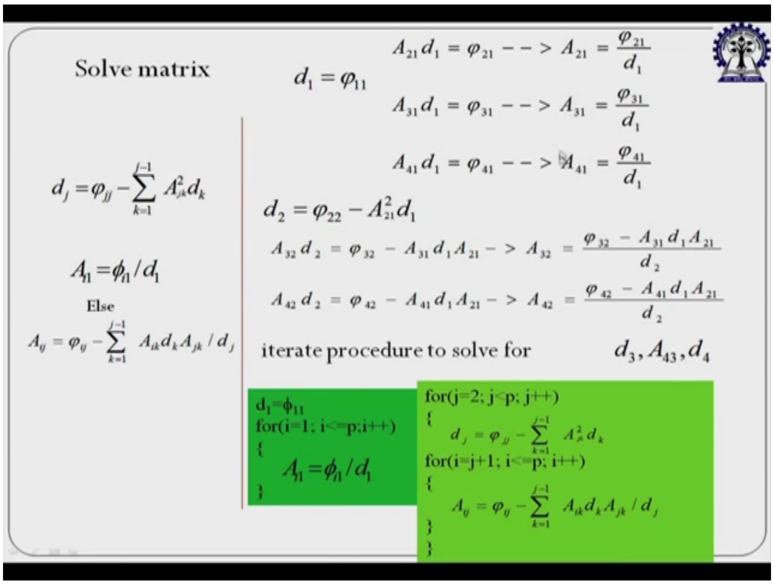

If you see that A 3 1; you do it phi 3 1 by d 1, A 4 1; phi 4 1, by d 1. Now, if I solve for d 2, it will come phi 2 2; this will phi; there is a some typing mistake. So, phi 2 2; minus A 2 1 square into d 1; this will be phi; all are phi this is phi. So, I can write down the program, I can generalize this d equation and generalize the A equation and write down that program. So, I can find out the d and A value.

Now, once I get the d and A value; I find out d and A value I get in terms of other d and A value. So, d 1 in terms of phi value; I get it, now if I see ADA transpose; alpha is equal to let us this one; instead of A; this one I write A into y. So, I can say D A transpose alpha; D A transpose alpha is equal to it is Y matrix.

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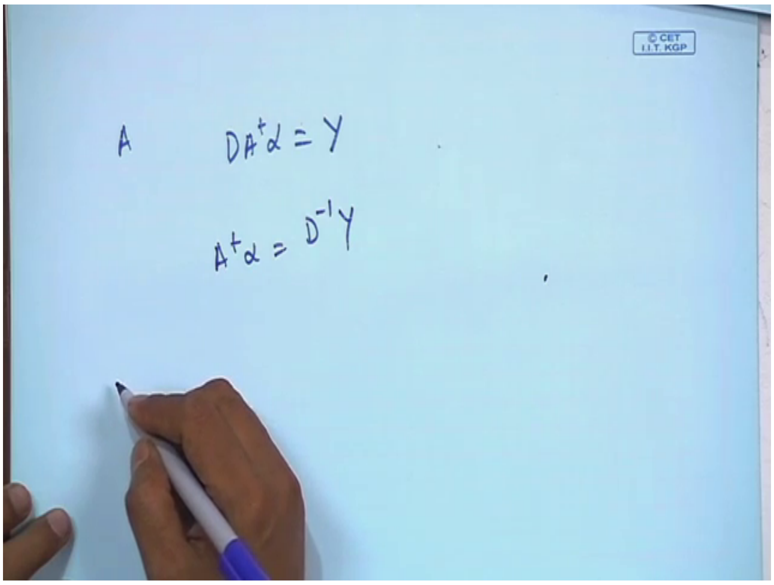

So, I can write down this one again; A 1, Y 1 and this one and then again I solve and generalize this one; for any order, this is the generalization; I can write down this matrix, I get Y value. Once I get that Y value; so, this is D A transpose alpha is equal to Y. So, I can say A transpose alpha is equal D inverse Y. So, D inverse I can easily calculate; then I can say alpha A transpose alpha is equal to D inverse Y. So, I put that A value; I put the alpha value matrix; D transpose and Y, then I can get that alpha 4, alpha 3, alpha 2, alpha 1. So, it is reverse order we are getting; we can write down the equation generalize program.

So, I can using this matrix decomposition technique; I also can calculate the alpha value; if I know the phi value. Because phi value is nothing but a phi ik is nothing but a this one using the window. Only difference is that in matrix; this covariance matrix at n equal to 0 this signal is not 0; I required the P number of previous sample. Because this restrict me; so, this is called autocorrelation methods. Then there is a other techniques also that is called lattice methods; that we will discuss in the next class.

Let us discuss in one thing which is there; which is important which is called a gain.

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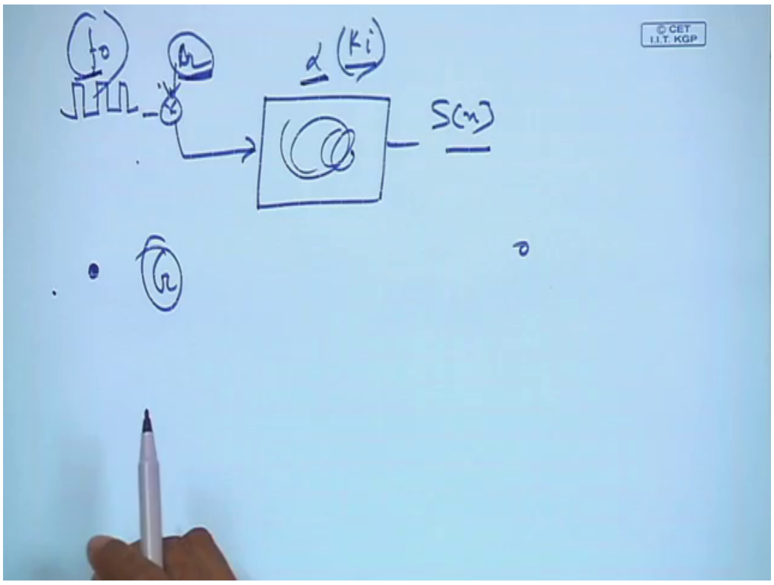

So, what I am saying in LPC analysis; we can say if I apply the; if I know the alpha value or K value, I can design this filter and if I know the voice signal is nothing but a impulse. So, I generate the impulse based on that F 0 value of the voice signal and multiply the gain; that is G and then if I can pass through this filter; I can generate the speech signal.

So, if it is LPC decoder; the simple LPC encoder and decoder. So, if I want to transmit this signal from this point to this point; using simple LPC encoding and decoding, what I want? I want to send that F 0 value; Z value, alpha value or K i value in the receiver end. And from F 0 value, Z value and alpha value; I can generate the speech signal. So, F 0 is extracted based on the; if I get the speech signal, I can extract that F 0 value.

So, next I can use K i value; I can extract that K i value using the autocorrelation technique or covariance technique. Then next I have to know the G value; what is the G value? Gain value. So, how do you calculate the g? If you see; I can calculate easily I can calculate the computational model gain; gain of the signal.

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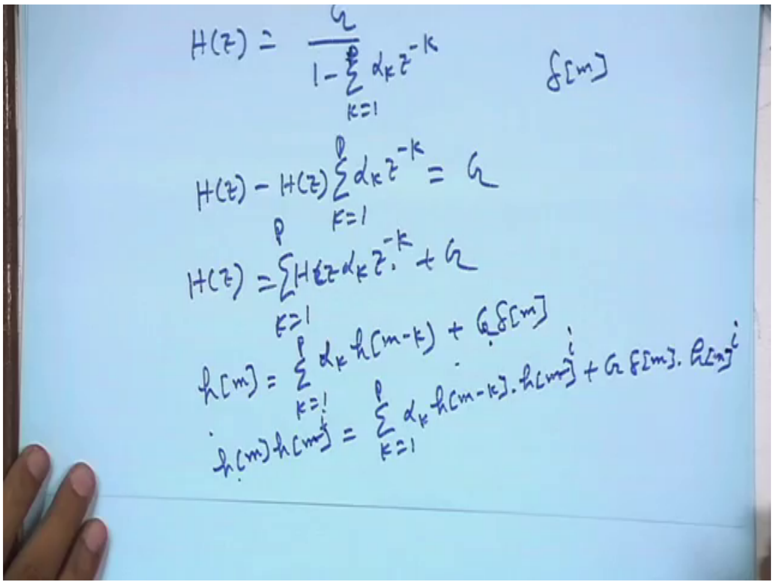

If you see H Z is nothing but a G by 1 minus K equal to 1 to P; alpha K Z to the power minus K. This is the LPC equation or I can say; in time domain if I multiply it this side H Z minus H Z into K equal to 1 to P; alpha K Z to the power minus K is equal to G.

So, I can say H Z is equal to H Z into K equal to 1 to P; alpha K, Z to the power minus K plus G. So, if it is H Z it is nothing but I can time domain h m is nothing but a K equal to 1 to P is alpha K h of m minus K; K sample delay plus G; if I say my input of the system is a delta function. So, I can say the G; G is nothing but a gain with a delta function at m equal to 0; only G exists with the gain.

Now, if I multiply both side h m multiply by h m. So, I can say K equal to 1 to P; alpha K h of m minus K into h m plus G del m into h m or not. So, if it is that; so, what is or m let us m minus i; multiply m minus i; m minus i, multiply both side by m minus i.

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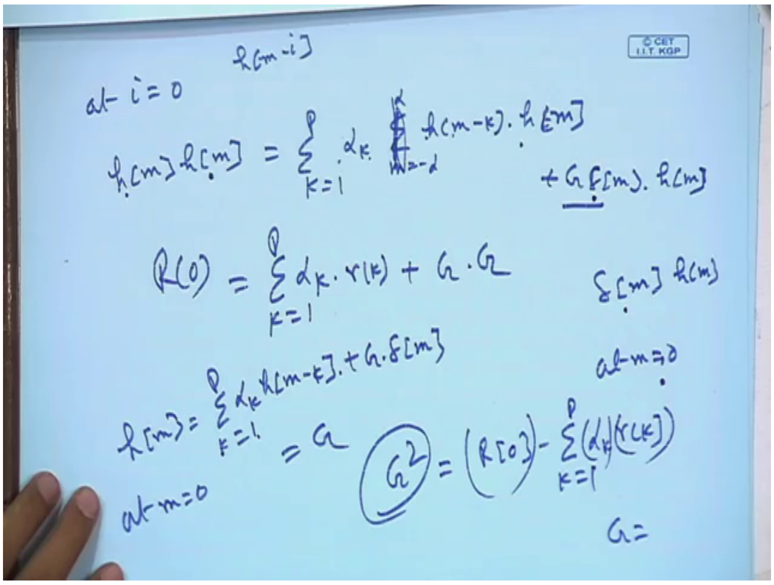

Now, at i equal to 0 h m into h m minus i is nothing but h m multiplied by h m; K equal to 1 to P; alpha K into m equal to minus infinity to infinity m minus i; I have taken minus infinity to infinity; h of m minus K into h m; forget about this part, alpha K into h of m minus K into h m; plus G of delta m into h m; now h of m into h of m is nothing but autocorrelation R 0; minus K equal to 1 to P alpha K; this is nothing but a R K plus this is G and what is del m and h m; at l equal to 0 del m is only exists at m equal to 0. So, I can say this h m is nothing but a K equal to 1 to P alpha K; h m minus K into plus G into del m.

Now, at m equal to 0 because del m is only exists at m equal to 0. So, m equal to 0, I can say K equal to 1; this is nothing but a G. So, I can say G into G; so, G square. So, I can say G square is nothing but a R 0 minus K equal to 1 to P; alpha K; rk, K equal to 1 to P alpha K; rk. If you know the alpha value; if I know the rk value, if I know the R 0 value; I can calculate G square; once I get the G square; I can calculate G value; is clear or not?

So, what about the speech segment I get; once I know the speech segment, I can calculate the alpha value, I can calculate the r value; r 0 and rk; all rk value, if the order of the predictor is P; then I can calculate r 1, r 2, r 3, r P and I can calculate alpha 1, alpha 2, alpha 3, alpha P. Then I calculate the G; once I get the G square, G is nothing but a root over of that things. So, model gain; I can easily compute using this theory.

So, next class we will discuss about that which is very important lattice modelling for LPC extraction and also LPC parameter that extraction also implementation of pole 0 filter all those kind of things; we will discuss. So, LPC synthesis because that is very important; how do you synthesize the speech signal at decoder; once I know the K value and alpha value.

Thank you.