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Intern rapport E-290
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A SYSTEM FOR DIRECTION EXTRACTING

by

A Flygansvær

Kjeller, 15 November 1978

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SUMMARY

This report contains the theoretical work that has been done in connection with a project, where the scope is to remote control direction finding (DF) systems. The content is limited to a system for calculating the azimuth out of signals from existing DF equipment.

Different types of signals that appear, and the advantages of using frequency analysis to these signals, are discussed. Two methods for realizing a frequency analysis are described. A system configuration is also described as to how a direction extraction might be achieved. The function of such a system was simulated. Some of the results are included and they show that the chosen system configuration is very promising, but this can not be verified before a total system has been built.

1 THE DIRECTION FINDER

The direction finder (DF) is connected to an antenna system, and is used for indicating the azimuth of an incoming electro-magnetic wave. The azimuth will be shown on a CRT-display. The DF-system covers a wide frequency range with a tuning capability in smallest steps of 100 Hz. The frequency can be tuned either manually or by digital remote control.

A simple block diagram of a direction finding system is shown in figure 1. There are three channels; AB, CD and HA. The signals in channel AB and CD come from a reducing goniometer and they determine the azimuth of an incoming electro-magnetic wave. The principle of this is shown in figure 2. In the goniometer the signals from the antenna system will be made into two voltages, U_{AB} and U_{CD}, which give the y and x coordinate of an incoming wave's direction.

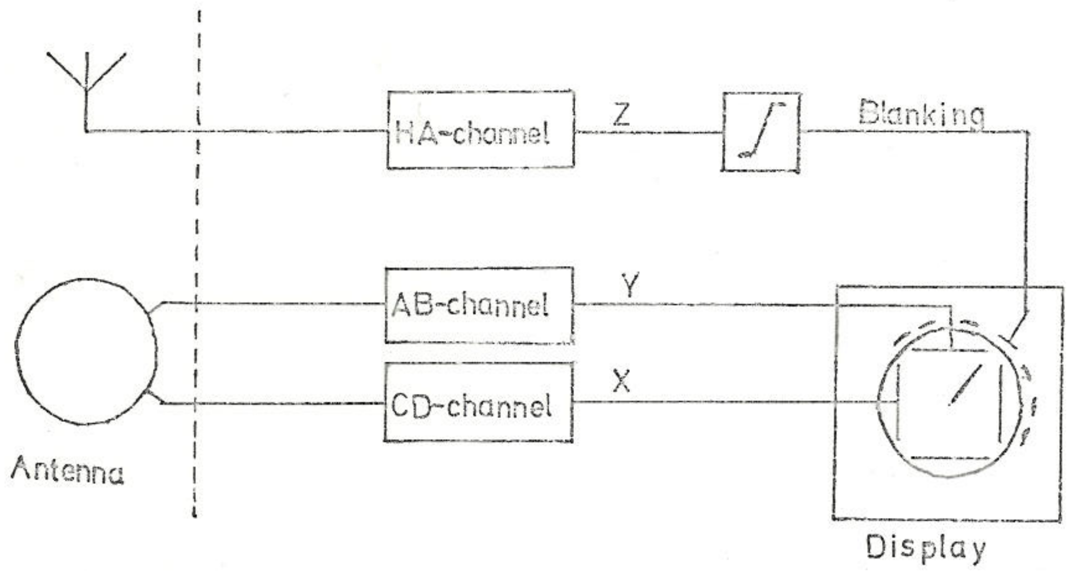

Figure 1 DF-system

The information in channel AB og CD will give both the correct and the opposite simultaneously. The signal in the third channel (HA) comes from an auxiliary antenna and is used to blank out one half of the trace on the CRT so that the correct direction may be found. It is also used to detect information carried by the incoming wave.

To obtain correct DF indications both phase and signal amplification of channel AB and CD must be equal. This is secured by automatic balancing at intervals of about 50 sec. Automatic gain control is used in all channels.

All the signals to the CRT are converted to about 62kHz.

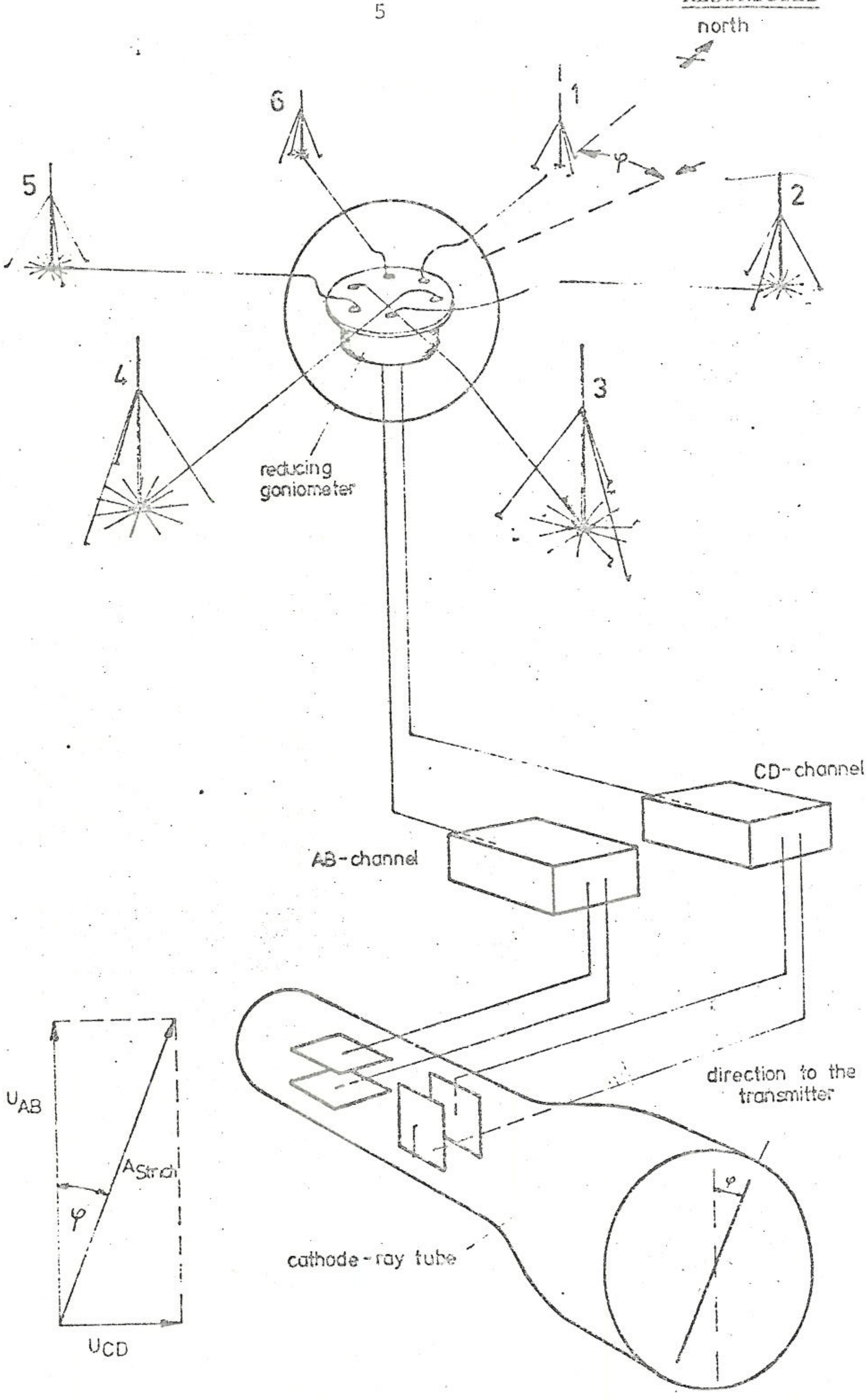

Figure 2 Indication principle of visual direction finder

2 THEORY ABOUT APPEARING SIGNALS

When an electro-magnetic wave with an arbitrary polarisation is received the directional antenna system will convert its direction into two signals which give the x and y coordinates:

Channel CD : $x = A \cdot \cos\theta \cdot \cos wt$ (1)

Channel AB : $y = A \cdot \sin\theta \cdot \cos wt$ (2)

In addition there is the blanking signal:

Channel HA : $z = k \cdot A \cdot \cos wt$ (3)

The angle θ indicate the azimuth and wt gives the frequency of the x, y and z signal. See figure 1. A is the amplitude of the incoming wave and k is a factor to compensate for not equal amplification in the third channel.

When only one wave is present the direction will be indicated by a straight line.

This can be shown by using eq. 1 and 2:

$$\frac{x}{y} = \frac{A \cdot \cos\theta \cdot \cos wt}{A \cdot \sin\theta \cdot \cos wt} = \frac{\cos\theta}{\sin\theta}$$

$$y = \tan\theta \cdot x \quad (4)$$

The picture becomes more complicated when several waves are received simultaneously. It is common to distinguish between two types of interference; coherent and incoherent. Coherent interference appears when more than one wave are present and they have equal frequency. With different frequencies the interference is incoherent.

2.1 Coherent interference

In the general case a coherent interference will consist of a sum of signals with the same frequency, but with different phases. The signals in the three channels will be of the form:

$$x = \sum_i A_i \cdot \cos\theta_i \cdot \cos(wt+\phi_i) \qquad (5)$$

$$y = \sum_i A_i \cdot \sin\theta_i \cdot \cos(wt+\phi_i) \qquad (6)$$

$$z = \sum_i A_i \cdot \cos(wt+\phi_i) \qquad (7)$$

The value of k is set to be one. These equations may be written on the form:

$$x = A_x \cdot \cos(wt+\phi_x) = (A_x \cdot \cos\phi_x) \cdot \cos wt - (A_x \sin\phi_x) \cdot \sin wt \qquad (8)$$

$$y = A_y \cdot \cos(wt+\phi_y) = (A_y \cdot \cos\phi_y) \cdot \cos wt - (A_y \cdot \sin\phi_y) \cdot \sin wt \qquad (9)$$

$$z = A_z \cdot \cos(wt+\phi_z) = (A_z \cdot \cos\phi_z) \cdot \cos wt - (A_z \cdot \sin\phi_z) \cdot \sin wt \qquad (10)$$

The x and y signal (Eq 8 and 9) will generate an ellipse on the CRT-display. To show this the phase difference, $\Delta\phi=\phi_x-\phi_y$, is inserted and eq 8 and 9 are combined to eliminate the parameter wt. This result in:

$$(A_y^2)x^2-(2A_xA_y \cdot \cos\Delta\phi) \cdot xy+(A_x^2)y^2=A_x^2A_y^2\sin^2\Delta\phi \qquad (11)$$

To find the direction of the ellipse eq 11 can be transformed into a new coordinate system given by the transformation:

$$\begin{bmatrix} x \\ y \end{bmatrix} = \begin{bmatrix} \cos\sigma & -\sin\sigma \\ \sin\sigma & \cos\sigma \end{bmatrix} \cdot \begin{bmatrix} x^1 \\ y^1 \end{bmatrix} \qquad (12)$$

The new ellipse-equation becomes:

$$y^{1^2}(A_x^2 \cdot \cos^2\sigma+A_xA_y \cdot \cos\Delta\phi \cdot \sin 2\sigma+A_y^2\sin^2\sigma)+$$

$$+x^1y^1(A_x^2\sin^2\sigma \div 2A_xA_y \cdot \cos\Delta\phi \cdot \cos 2\sigma-A_y^2 \sin 2\sigma)+$$

$$+x^{1^2}(A_x^2 \cdot \sin^2\sigma - A_x A_y \cdot \cos\Delta\phi \cdot \sin 2\sigma + A_y^2 \cos^2\sigma) =$$

$$= A_x^2 \cdot A_y^2 \cdot \sin^2\Delta\phi$$

When the new coordinate system lies in the ellipse - axis the x^1y^1-term has to disappear, and the ellipse - direction can be found:

$$\tan 2\sigma = \frac{2A_x \cdot A_y \cdot \cos\Delta\phi}{A_x^2 - A_y^2} \qquad (13)$$

An other interesting parameter is the ratio between the ellipse's axis. The equation for this is:

$$T = \sqrt{\frac{1 \pm \sqrt{1 - \frac{4A_x^2A_y^2 \sin^2\Delta\phi}{(A_x^2 + A_y^2)^2}}}{1 \mp \sqrt{1 - \frac{4A_x^2A_y^2 \sin^2\Delta\phi}{(A_x^2 + A_y^2)^2}}}} \qquad (14)$$

If σ shall be the angle of the great axis then the following condition of eq 14 must be satisfied : $T \geq 1$

This leads to :

$$A_x > A_y : -45^\circ < \sigma < 45^\circ \text{ or } 135^\circ < \sigma < 225^\circ$$
$$A_x < A_y : \ 45^\circ < \sigma < 135^\circ \text{ or } 225^\circ < \sigma < 360^\circ \qquad (15)$$

The greatest possibility for coherent interference is because of reflections. There might be one or more reflections present. In general it is not possible to determine the directions of the separate components. An infinitely combination of vectors may generate the same ellipse. If the assumption is made that the direct wave is much stronger than the reflections then the direction of the ellipse will be a good estimate of the wanted azimuth. The ellipse's parameters can be evaluated using eq 13 and 14. The direction gives two alternatives 180° opposite to each other. The phase of the z-signal must then be used to eliminate the wrong one. This can be done in the following way: The ellipse has two top-points defined as the points where the great axis crosses the ellipse. From eq 8 and 9 the parameter wt can be evaluated to give these points. Inserting the evaluated wt's in eq 10 will give a positive and a negative value for z. The wt which give the positive value is the right one. This value is then used to determine which top-point the direction shall pass through.

2.1.1 An electro-magnetic wave with one reflection

With the assumption that there is only one reflection the signals x, y and z can be derived from eq 5, 6 and7:

$$x = A\cdot\cos\theta_1\cdot\cos(wt+\phi_1)+B\cdot\cos\theta_2\cdot\cos(wt+\phi_2) \quad (16)$$

$$y = A\cdot\sin\theta_1\cdot\cos(wt+\phi_1)+B\cdot\sin\theta_2\cdot\cos(wt+\phi_2) \quad (17)$$

$$z = kA\cdot\cos(wt+\phi_1)+kB\cdot\cos(wt+\phi_2) \quad (18)$$

Figure 3 illustrates the signals. The direct wave has amplitude A and an angle θ_1 with the x-axis.

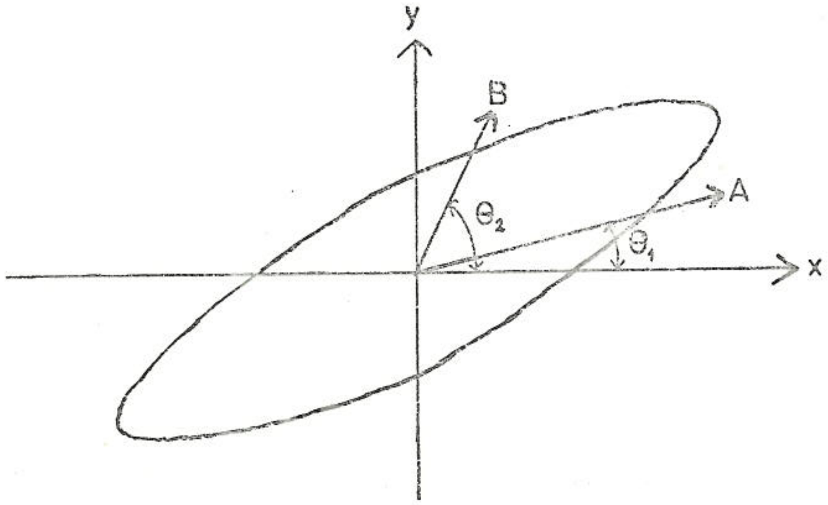

Figure 3 Signal with reflection

The reflected wave has amplitude B and an angle θ_2 with the x-axis. The constant k is used to compensate for the difference in amplification in the third channel.

The primary thing of interest is to determine the value of θ_1. This can be done if the constant k is known. The method will be discussed in the following.

The signals that can be extracted from the DF-system are of the form (eq 8, 9 and 10):

$$x = Q \cdot \cos wt + P \cdot \sin wt \quad (19)$$

$$y = S \cdot \cos wt + R \cdot \sin wt \quad (20)$$

$$z = M \cdot \cos wt + L \cdot \sin wt \quad (21)$$

where

$$Q = A_x \cdot \cos\phi_x, \quad P = -A_x \cdot \sin\phi_x$$

$$S = A_y \cdot \cos\phi_y, \quad R = -A_y \cdot \sin\phi_y$$

$$M = A_z \cdot \cos\phi_z, \quad L = -A_z \cdot \sin\phi_z$$

These parameters can be regarded as known. The relation between the known parameters and the unknown in eq 16, 17 and 18 is:

$$A\cdot\cos\theta_1\cdot\cos\phi_1+B\cdot\cos\theta_2\cdot\cos\phi_2 = Q \quad (22)$$

$$A\cdot\cos\theta_1\cdot\sin\phi_1+B\cdot\cos\theta_2\cdot\sin\phi_2 = -P \quad (23)$$

$$A\cdot\sin\theta_1\cdot\cos\phi_1+B\cdot\sin\theta_2\cdot\cos\phi_2 = S \quad (24)$$

$$A\cdot\sin\theta_1\cdot\sin\phi_1+B\cdot\sin\theta_2\cdot\sin\phi_2 = -R \quad (25)$$

$$kA\cos\phi_1 + kB\cos\phi_2 = M \quad (26)$$

$$kA\cdot\sin\phi_1 + kB\sin\phi_2 = -L \quad (27)$$

Solving these equations for θ_1 and θ_2 give:

$$(MP-LQ)\sin\theta_1 + (LS-MR)\cos\theta_1 = (PS-QR)k \quad (28)$$

$$(MP-LQ)\sin\theta_2 + (LS-MR)\cos\theta_2 = (PS-QR)k \quad (29)$$

These two equations give the same solutions:

$$\theta_{1,2} = 2\arctan\frac{E\pm\sqrt{E^2+F^2-k^2G^2}}{F+kG} \quad (30)$$

$E = MP-LQ, \quad F = LS-MR, \quad G = PS-QR$

Eq 30 gives two possibilities of which one is the solution for θ_1 and the other for θ_2.

The following eq can be used to determine which is which:

$$\frac{B^2}{A^2} = \frac{(S\cdot\cos\theta_1-Q\sin\theta_1)^2 + (R\cdot\cos\theta_1-P\sin\theta_1)^2}{(Q\cdot\sin\theta_2-S\cdot\cos\theta_2)^2 + (P\sin\theta_2-R\cos\theta_2)^2} \quad (31)$$

With the assumption that the direct wave is stronger than the reflected the following condition must be satisfied : $A > B$.

The solutions for θ_1 and θ_2 must then be chosen to meet this condition.

Some special consideration must be taken when the signals x, y and z have zero or 180° phase difference. This will result in:

$E = F = G = 0$ and

eq 30 is no longer valid. A possible solution would be to assume that there is no reflection which give:

$$\theta_1 = \arctan\left(\frac{R}{P}\right) \text{ if } R \neq 0 \text{ and } P \neq 0 \tag{32}$$

$$(\theta_1 = \theta_1 + 180^\circ \text{ if } P\,L < 0)$$

or

$$\theta_1 = \arctan\left(\frac{S}{Q}\right) \text{ if } S \neq 0 \text{ and } Q \neq 0 \tag{33}$$

$$(\theta_1 = \theta_1 + 180 \text{ if } QM < 0)$$

If there is no reflection the following condition must be satisfied:

$$Q^2+P^2+S^2+R^2 = \frac{1}{K^2}\,(M^2+L^2) \tag{34}$$

The described method uses both phase and amplitude of the signal in the third channel. There is some uncertainty concerning the z-signal because it is not treated equal in the DF-system relative to the x- and y-signal. It is therefore uncertain how usable the method is.

2.2 Incoherent interference

The general case of incoherent interference is:

$$x = \sum_i A_i \cdot \cos\theta_i \cdot \cos(w_i t+\phi_1) \tag{35}$$

$$y = \sum_i A_i \cdot \sin\theta_i \cdot \cos(w_i t+\phi_i) \tag{36}$$

$$z = \sum_i A_i \cdot \cos(w_i t+\phi_i) \tag{37}$$

If each of the waves can be separated in frequency then the different frequencies can be treated separately. In the next chapter methods of doing this will be discussed.

A combination of coherent and incoherent interference may also occur.

3 FREQUENCY ANALYSIS

Frequency domain representation of the signals have several advantages to our signal processing tasks; separating signal-components with different frquencies, obtain better signal/noise ratio, extracting information. To get this representation the signals have to be transformed from the time domain as illuatrated in figure 4.

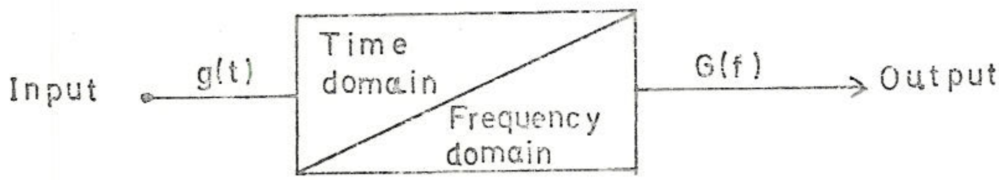

Figure 4 Transformation

In the following two methods will be described as how to obtain a frequency transformation. The main differences in the two methods are that one uses digital signals and an algorithm called the Fast Fourier Transform (FFT), while the other uses analog signals, discrete in time, and an algorithm called Chirp Z Transform (CZT). A common feature of both methods is that they make use of the discrete Fourier Transform (DFT). The DFT of a sequence $\{f(n)\}$ is given by:

$$F(k) = \sum_{n=o}^{N-1} f(n) \cdot e^{-j(\frac{2\pi}{N})nk} \qquad k = 0,1,\cdots, N-1 \quad (38)$$

The set of N complex numbers, $\{F(k)\}$, is called the discrete Fourier transform of the input sequence. The DFT is very similar to the Fourier integral and Fourier series. Some properties of the DFT are:

- The analysed frequency band in a normal situation extends from zero to the Nyquist frequency (one-half of the sample frequency).
- The input sequence is in general complex.

- The resolution in frequency is approximately 1/N of the sample frequency.
- When the input sequence is real the output sequence will contain redundancy.
- The numbers in the transformed sequence are in general larger than the numbers in the input sequence.

3.1 The FFT - processor

The computation of the DFT can be done by using an algorithm generally known as the Fast Fourier Transform (FFT). The main property of FFT is that the number of computations is drastically reduced compared to a straight forward use of the DFT.

Although the FFT consists of a whole class of algorithms only the algorithm called "Radix 2 decimation-in-time" will be taken into consideration. The algorithm for this is:

$$F_k = X_1(k) + W_n^k \cdot X_2(k)$$

$$F_{k+\frac{N}{2}} = X_1(k) - W_N^k \cdot X_2(k), \qquad k=0,1,\ldots,\tfrac{N}{2}-1 \qquad (39)$$

$$W_N = \exp(-j\tfrac{2\pi}{N})$$

$$X_1 = \sum_{n=o}^{\frac{N}{2}-1} X(2n) \cdot W_{\frac{N}{2}}^{nk}$$

$$X_2 = \sum_{n=o}^{\frac{N}{2}-1} X(2n+1) \cdot W_{\frac{N}{2}}^{nk} \qquad n=0,1,\ldots,\tfrac{N}{2}-1$$

$$W_{\frac{N}{2}} = W_N^2$$

Figure 5 shows a flow graph pictorial representation of an 8 point FFT. The algorithm is built up of an elementary configuration referred to as the butterfly.

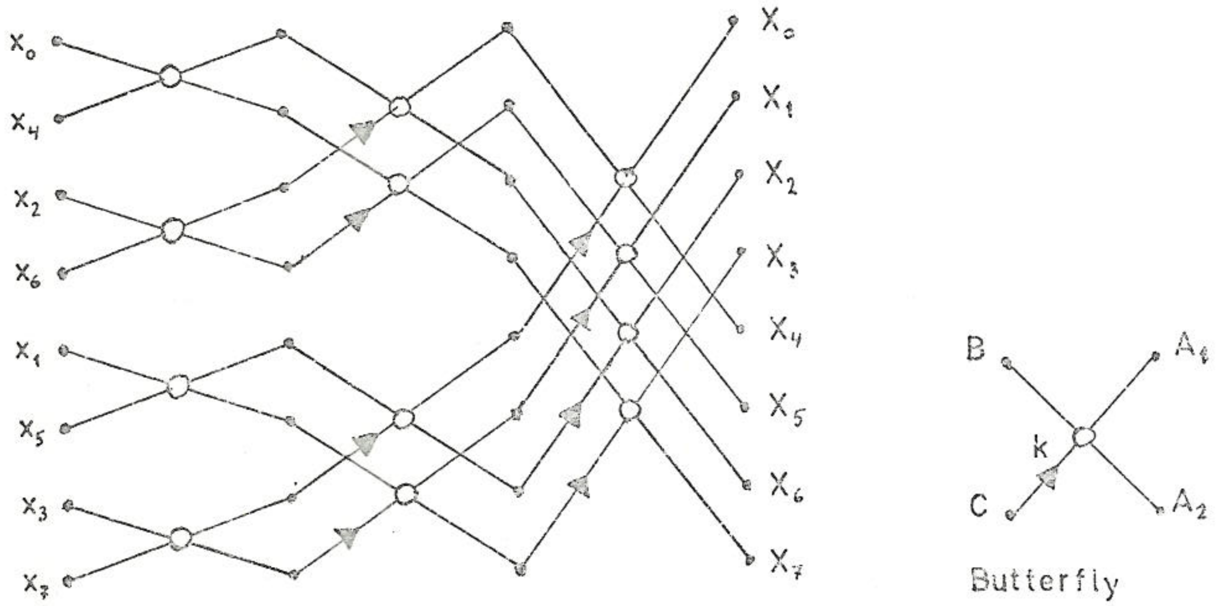

Figure 5 Flow graph

The configuration of this is:

$$A_1 = B+k\cdot C$$
$$A_2 = B-k\cdot C \qquad (40)$$

This suggests that A_1 and A_2 should be computed together by first forming the product $k\cdot C$ then adding and substracting the result from B.

This must be done N/2 times to perform the whole transformation.

Several methods could be used to implement the FFT-precessor. A relative simple method is to use a multiplier/accumulator as the arithmetic unit. A suggestion as to how this can be done is shown in figure 6.

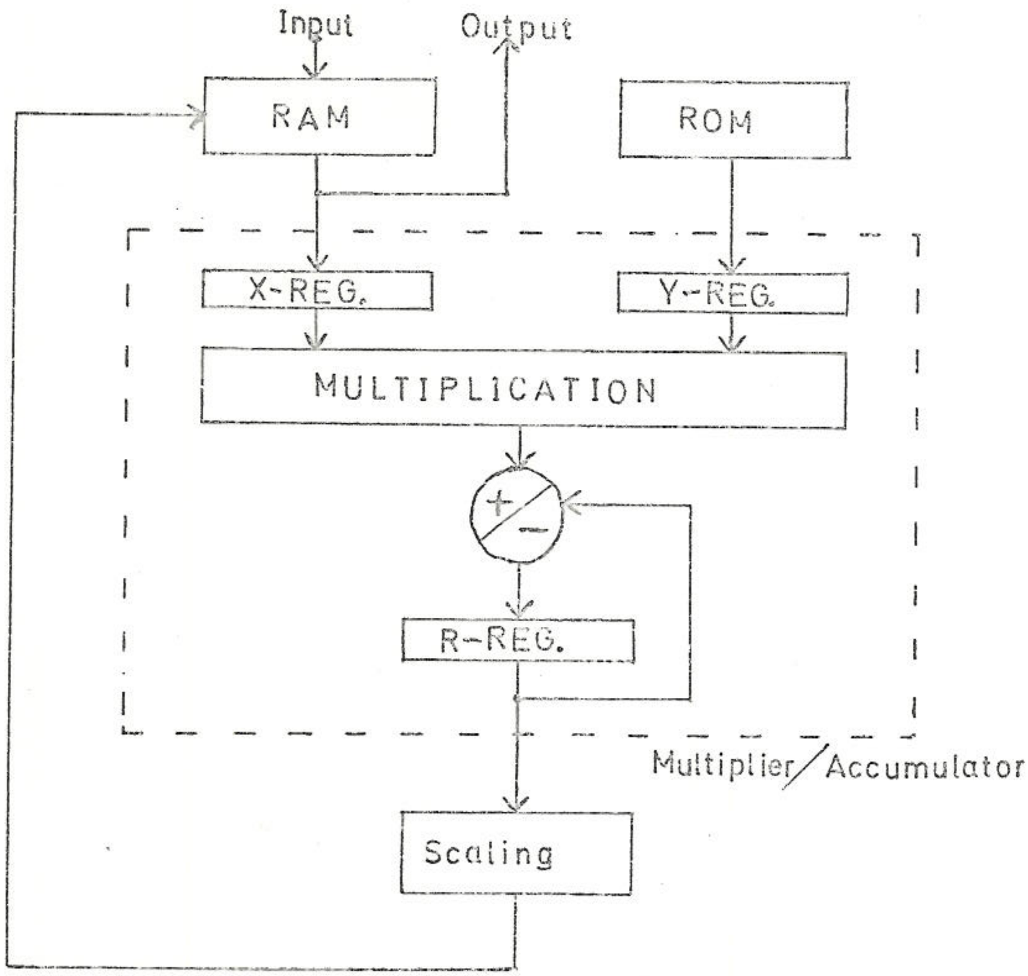

Figure 6 FFT - processor

The multiplier/accumulator work with digital numbers. The wordlength can be either 8, 12 or 16 bits. For a 12 bit unit one multiplication and accumulation can be done in 175 n sec with a TDC 1003j from TRW.

The input sequence of N numbers are read into the Random Access Memory (RAM). All the constants that are necessary are stored in Read Only Memory (ROM)

How it would work (ref to eq 40):

T_1 : Fetch C and constant k to x- and y-reg.

T_2 : Perform multiplication and store result in R-reg. Fetch B and constant "1" to x- and y-reg.

T_3 : Perform multiplication and add result to content of R-reg.

T_4 : Transfer content of R-reg. to memory. Perform same multiplication and substract content of R-reg from result.

T_5 : Perform same multiplication and substract content of R-reg from result.

T_6 : Transfer content of R-reg. to memory.

Six steps are necessary to perform a butterfly with real data. With complex data more steps are needed.

A scaling is indicated in figure 6. This is usually necessary to prevent overflow due to the fact that the DFT-sequence in general is larger than the input sequence.

The control unit which is necessary will not be discussed, but the main property of this would be to handle the data flow and control the multiplier/accumulator as efficiently as possible without becomming to complex.

3.2 The CZT-processor

A rearranging of eq 38 can be done by replacing the factor 2nk with the equivalent:

$$2nk = n^2+k^2 - (k-n)^2$$

Eq 38 then becomes:

$$F(k) = \sum_{n=o}^{N-1} [f(n)\cdot \exp(-j\frac{\pi n^2}{N})\cdot \exp(j\frac{\pi (k-n)^2}{N})]\cdot \exp(-j\frac{\pi k^2}{N}) \qquad (41)$$

There are three operations indicated by eq 41:

1 Multiply each corresponding term of the input discrete-time series, f(n), by the complex factor, $\exp(-j\frac{\pi n^2}{N})$, and obtain a new sequence.

2 Perform a discrete convolution between this new sequence and the sequence $\exp(j\frac{\pi (k-n)^2}{N})$.

3 Multiply the resulting output sequence by the final factor $\exp(-j\frac{\pi k^2}{N})$.

A system based on the above relationships is designated by the term Chirp Z Transform (CZT). A complete DFT based on the CZT algorithm is shown in block form in figure 7.

The CZT algorithm alone has no special advantage over other FFT methods for digital implementation, but a Charged Coupled Device unit (CCD) which performs the convolution part of the CZT has made it attractive to use. The CCD contain four 512-tap split-electrode transversal filters, with two sine chirps and two cosine chirps.

A DFT-processor that use the CCD can be made as figure 8 shows in blockform.

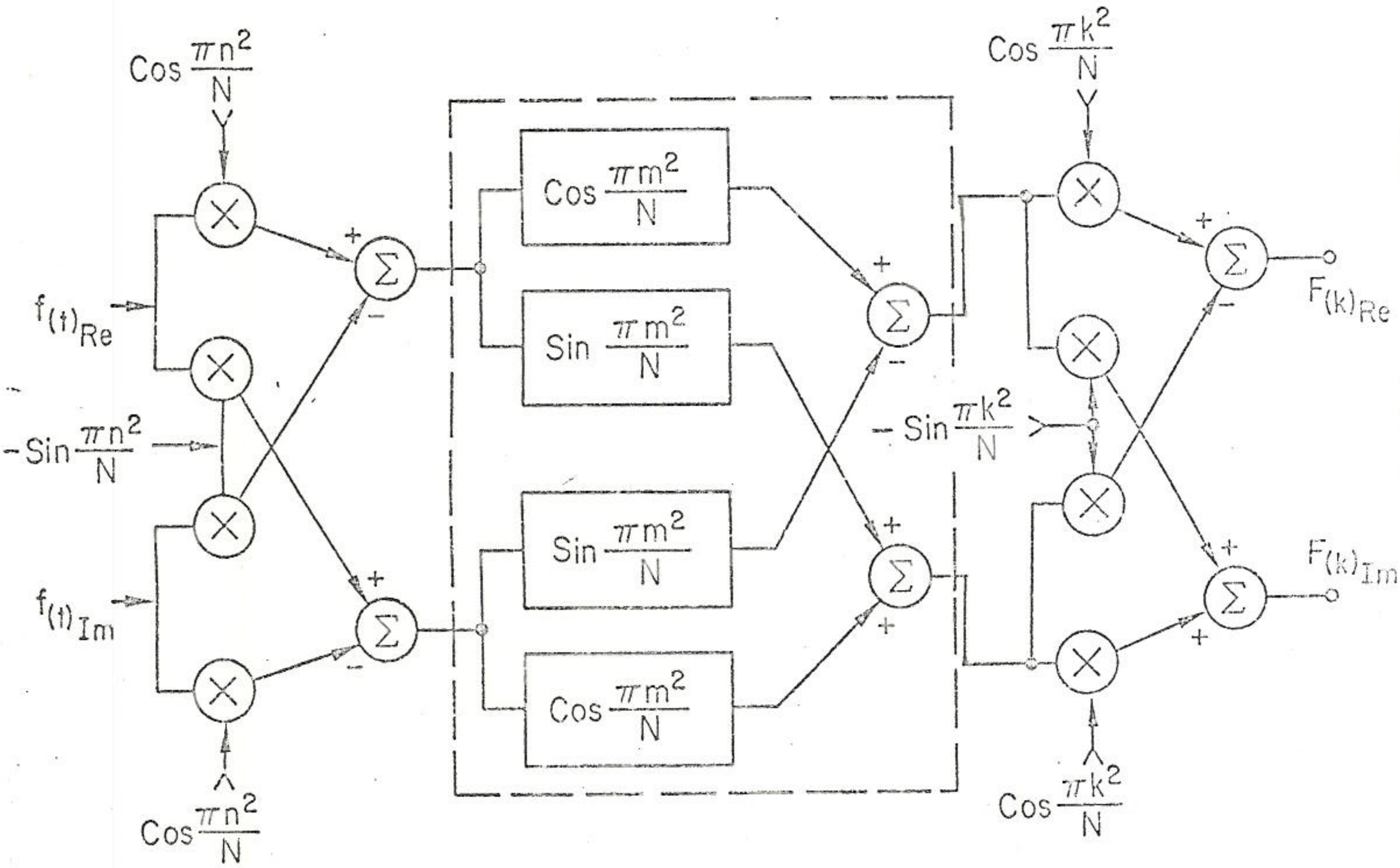

Figure 7 CZT based DFT

Some simplification is done compared to figure 7. When the input is purely real two of the input multipliers may be deleted. In addition it is shown how the magnitude of the power spectrum may be obtained without having to multiply the output sequence by the factor $\exp(-j\frac{\pi k^2}{N})$.

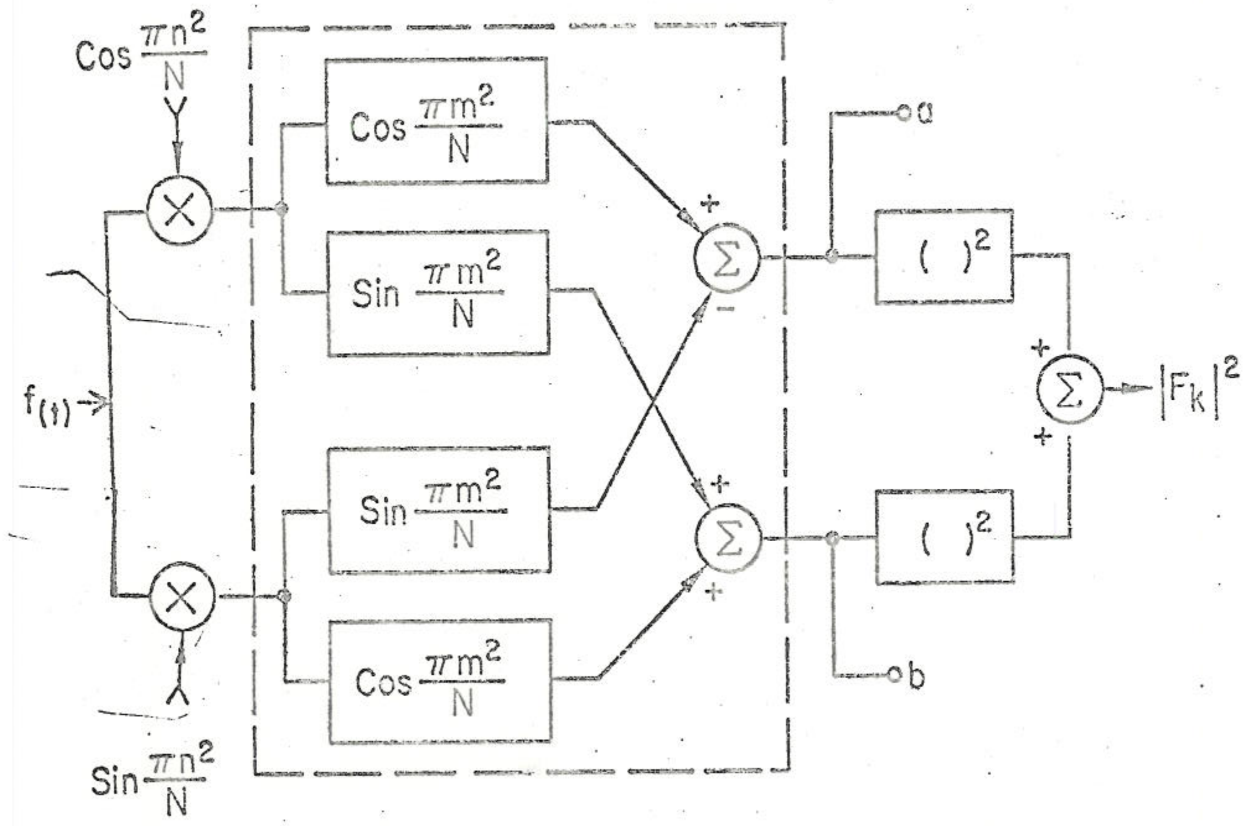

Figure 8 CZT-processor

This is due to the fact that the factor will only effect the phase of the frequency spectrum and not the amplitude.

3.3 Frequency Zoom

As described earlier, when using DFT the analysed frequency band in a normal situation extends from zero to the Nyquist frequency with a resolution of approximately 1/N of the sample frequency. The signals from the DF-systems to be analysed have a narrow bandwidth around 62kHz (with center frequency of 62kHz). Since there are no frequency components outside this band it is not necessary to analyse the frequency band from zero, which would give a bad resolution.

A method will now be described as how to use the DFT and analyse the desired frequency band.

If one has a signal, $x(t)$, with a frequency band as shown in figure 9 this can be sampled with a samplings function of the form:

$$s(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_o) \tag{42}$$

The sampled signal will be:

$$X_1(t) = X(t) \cdot S(t) \tag{43}$$

The frequency spectrum of the samplingsfunction is:

$$S(j\omega) = \frac{2\pi}{T_o} \sum_{k=-\infty}^{\infty} \delta(\omega-k\omega_o) \tag{44}$$

and this is shown in figure 10. The frequency spectrum of the sampled signal can then be evaluated and becomes:

$$X_1(j\omega) = \frac{1}{T_o} \Sigma\, X(j\omega-jk\omega_o) \tag{45}$$

This function is periodic with repeatings in a multiple of the sampling frequency, ω_o. See figure 11. The figure also shows why the sampling frequency has to be at least twice as high as the highest frequency in the signal, $X(j\omega)$, to avoid aliasing. The sampled signal can be used to calculate the DFT of the original signal, $x(t)$. The calculated band will be from $-\omega_o/2$ to $\omega_o/2$, symmetric about zero, as shown in figure 12. Because of the symmetry the only band of interest is from zero to $\omega_o/2$.

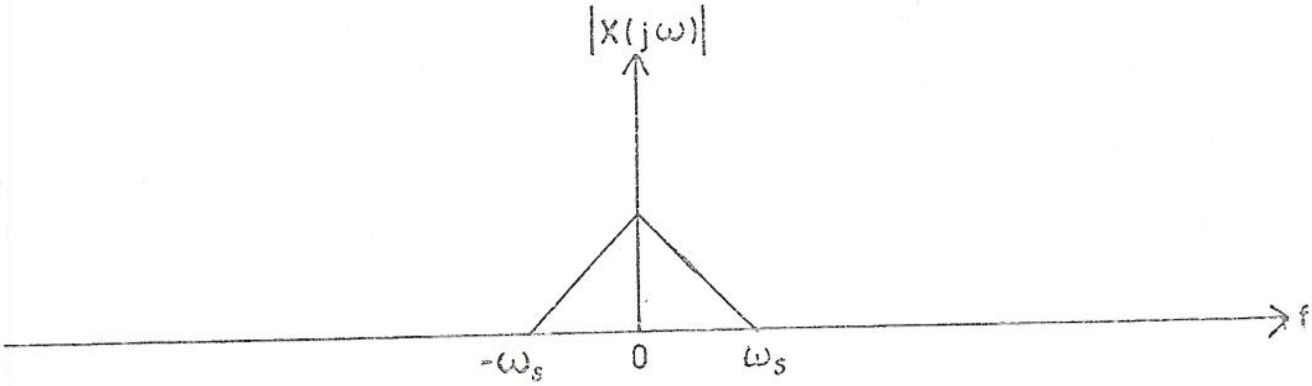

Figure 9 Frequency spectrum

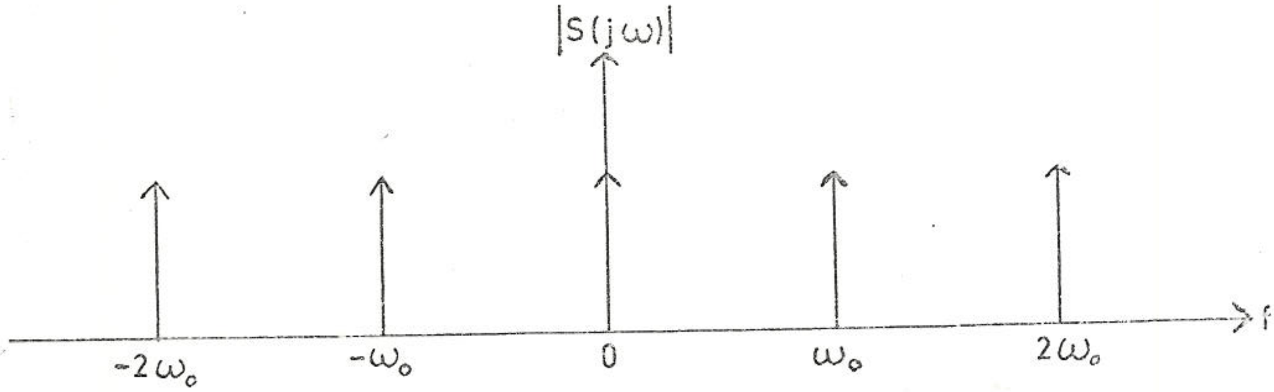

Figure 10 Samlingsfunction

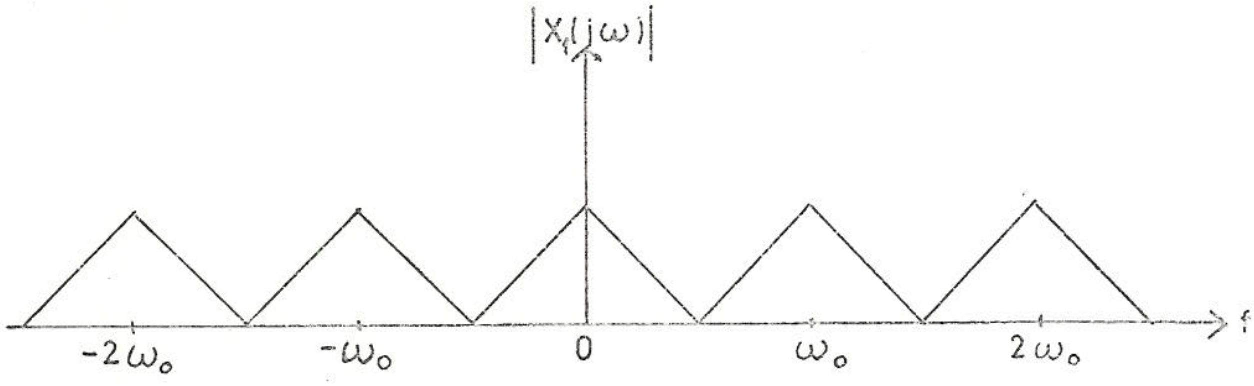

Figure 11 Frequency spectrum of sampled signal

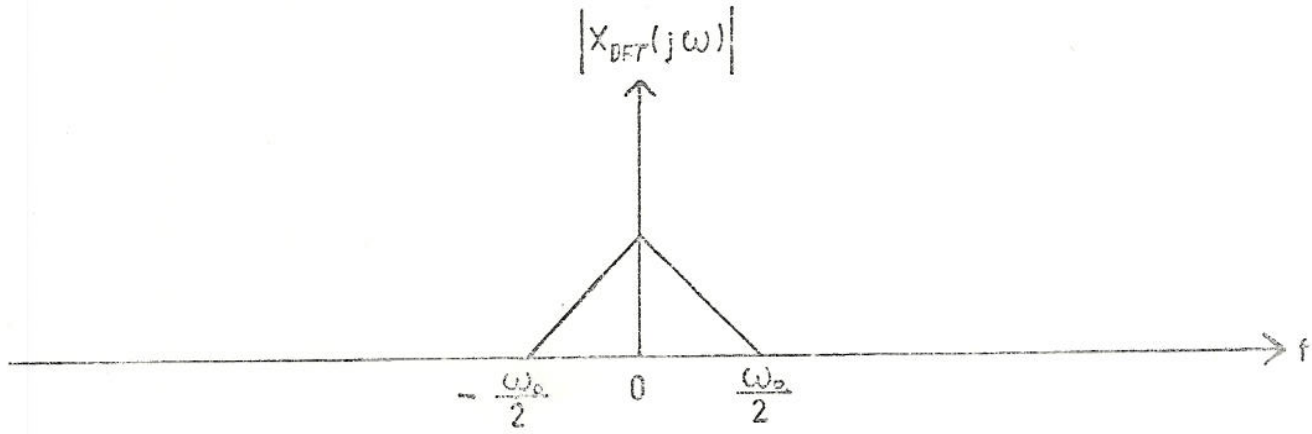

Figure 12 Analysed frequency band

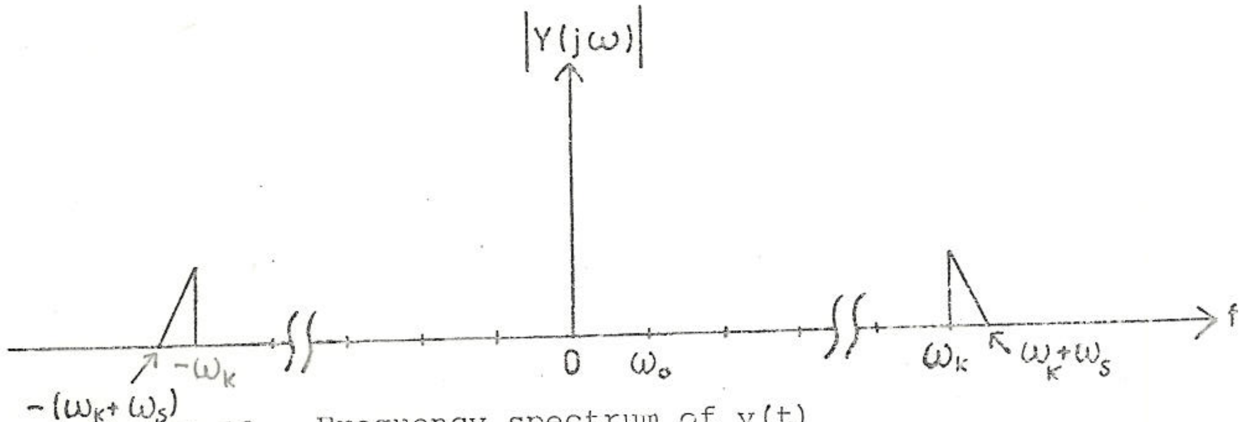

Figure 13 Frequency spectrum of y(t)

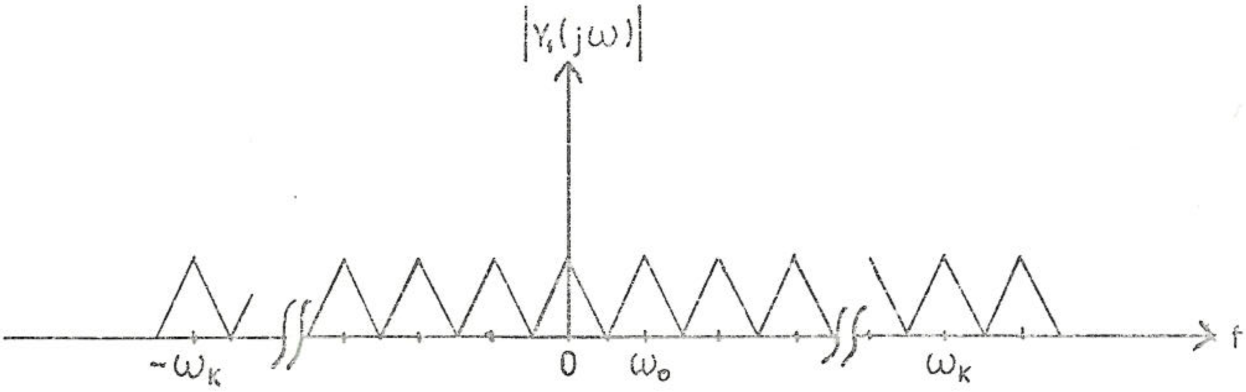

Figure 14 The frequency spectrum of y(t) after sampling

If there is a signal y(t) instead of x(t) with the frequency spectrum:

$$Y(j\omega) = X(j\omega - j\omega_k), \quad \omega_k = n \cdot \omega_o \qquad (46)$$

this will result in a spectrum as shown in figure 13. If this signal is sampled with the function in eq 42 the sampled signal becomes:

$$Y_1 (j\omega) = \frac{1}{T_o} \sum_{k=-\infty}^{\infty} Y(j\omega - jk\omega_o) \qquad (47)$$

$$Y_1(jw) = \frac{1}{T_o} \sum_{k=-\infty}^{\infty} X(jw - jw_k - jkw_o) = \frac{1}{T_o} \sum_{k=-\infty}^{\infty} X(jw \div j(n+k)w_o) \qquad (48)$$

$$Y_1(j\omega) = \frac{1}{T_o} \sum_{m=-\infty}^{\infty} X(j\omega - jm \cdot \omega_o) \text{ where } m=n+k \qquad (49)$$

Eq 49 is equal to 45. The spectrum of $Y_1(j\omega)$ is shown in figure 14. Since the signals x(t) and y(t) give an equal sampled signal under the given conditions the DFT of the two signals can not be distinquished. The DFT of y(t) will therefore be equal to the DFT of x(t).

The effect of the method (Frequency Zoom) is to move the frequency band, from some center frequency, down to a band around Zero, and do a transform on this low frequency signal, just by reducing the sampling frequency. There must be no signals outside the band that is to be analysed. This would create aliasing.

The way to use the method is to sample the signal with a frequency that is: $\omega_o = \omega_k/n$, where n is a positiv integer and ω_k is the frequency at one end of the original band to be analysed. ω_k can either be the lower limit, as show in figure 13, or the upper limit. $\omega_o \geq 2 \cdot \omega_S$.

4 DIRECTION EXTRACTING

From the DF-system three signals can be extracted. These signals can be used to calculate the direction of an incoming electro-magnetic wave. The necessary features of the system which shall calculate the direction can be summarized as:

- Fetch the three signals from the DF-system.
- Transform these signals into the frequency domain.
- Extract the information from the frequency representation.
- Calculate the direction of an incoming wave.

4.1 Simulating

In order to investigate the function of a system as mentioned above, simulations were done. Two different types of signals were used in the simulations:

1 Self generated signals.
2 Tape-recorded signals from a DF-system.

All the frequency transformation and information extracting was done by using a 5420 HP-frequency analyser. The parameters of interest were the amplitudes of the three signals (A_x, A_y, A_z) and the phase-differences between the z-signal and the two others ($\Theta_x - \Theta_z$, $\Theta_y - \Theta_z$). Calculations were done on a NORD-10 computer.

4.1.1 Self generated signals

A special circuity was built to generate the necessary signals to simulated a wave with one reflection. All the parameters could be set separately for the three channels so that the directions of the direct and reflected wave were known. A block scheme of the circuity in shown in fig 15.

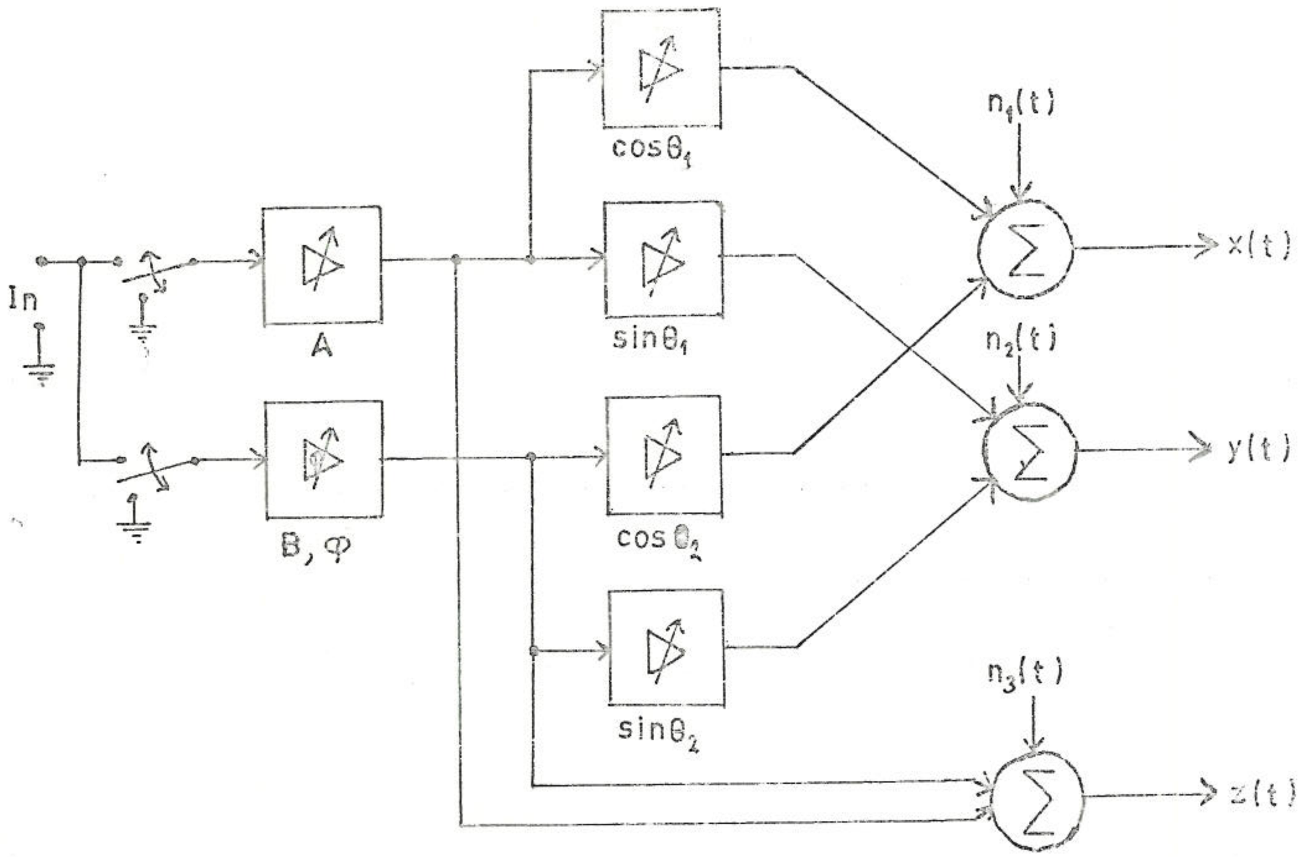

Figure 15 Circuit for generating signal with reflection.

The equations for the outputs are:

$$x=A^{\cdot}\cos\theta_1\cos wt+B^{\cdot}\cos\theta_2{}^{\cdot}\cos(wt+\phi)+n_1(t) \qquad (50)$$
$$y=A^{\cdot}\sin\theta_1\cos wt+B^{\cdot}\sin\theta_2{}^{\cdot}\cos(wt+\phi)+n_2(t) \qquad (51)$$
$$z=A^{\cdot}\cos wt+B^{\cdot}\cos(wt+\phi)+n_3t \qquad (52)$$

$n_1(t)$, $n_2(t)$ and $n_3(t)$ indicate noise which could be added to each of the three channels so that the signal/noise ratio could be varied.

The frequency of the signals was chosen to be 10kHz. By use of the HP-analyser the parameters of interest were extracted from the signals. The analyser could also be used to do averaging.

Measurements were done with three different settings which are shown in table 1. The direction, σ, of the ellipse is also included. The measurements are shown in table 2 and 3. From the measurements the direction, θ_1, is calculated together with the ellipse-parameters, σ and T, where T is the ratio between the axis. The equations for calculating θ_1, σ and T is described in 2.1. The measurements in table 3 were done with much noise which means a signal/noise ratio of about 2dB. Measurements were also done with less noise which gave similar results.

Since all the signals were stable the results should be independent of time, but measurements showed that there were introduced some errors in the frequency transformation, which gave a certain variation on the results of the same signals from time to time. In spite of this the results show that the estimation of the ellipse's direction is very good, with little variation, even when much noise were added. The direction, θ_1 is also good, with a little more variation, but this is due to the fact that in the calculation of θ_1 the z-signal is also used, and this introduces more error sources.

Setting	A	cos θ_1	sin θ_1	B	cos θ_2	sin θ_2	θ	σ
1	3	0,87	0,5	1	0,17	0,98	-60°	$38,1^\circ$
2	3	0,87	0,5	0	0,17	0,98	-60°	$29,9^\circ$
3	5	0,87	0,5	1	0,71	0,17	-30°	$32,2^\circ$

Table 1 Settings

	Measure-results					Calculations		
Setting	$\theta_x - \theta_z$	$\theta_y - \theta_z$	A_x	A_y	A_z	θ_1	σ	T
1	$9,9^\circ$	$-9,8^\circ$	949	745	1251,9	$29,3^\circ$	$37,7^\circ$	0,17
2	$-0,5^\circ$	$-0,5^\circ$	1012,8	569	1150,8	$29,3^\circ$	$29,3^\circ$	0,0
3	$0,9^\circ$	$-2,4^\circ$	1868,1	1188,5	2193,8	$30,5^\circ$	$32,5^\circ$	0,03

Table 2 Signals without noise.

		Measure-results					Calculations		
Setting	Number of averages	$\theta_x-\theta_z$	$\theta_y-\theta_z$	A_x	A_y	A_z	θ_1	σ	T
1	1	11,2°	-15,4°	999,4	836,6	1327,7	24,0°	39,4°	0,23
	5	9,6°	-9,8°	938,7	773,6	1276,8	28,0°	39,2°	0,17
	10	11,2°	-9,0°	930,8	758,3	1259,8	30,2°	38,8°	0,17
2	1	5,6°	5,7°	1133,9	644,6	1158	29,5°	29,6°	0,00
	5	-0,8°	-2,0°	1011,4	566,9	1072,2	27,1°	29,3°	0,01
	10	1,6°	-2,6°	999,5	550,-	1103,8	35,2°	28,8°	0,03
3	1	-5,2°	-2,3°	1805,8	1100	2301,3	30,0°	31,3°	0,02
	5	-1,0°	-2,6°	1870,9	1178,2	2127,1	31,0°	32,2°	0,01
	10	-1,7°	-5,4°	1866,8	1175,7	2116,0	30,7°	32,2°	0,03

Table 3. Signals with much noise

4.1.2 Tape-recorded signals

From a DF-system som signals were recorded using an instrumental tape-recorder.

The signals had a frequency about 62 kHz. The HP-frequency analyser that was used had only a frequency range up to 25 kHz. It was therefore necessary to convert the signals in frequency. This was done by reducing the speed of the tape-recorder to the fouth, which meant that the frequency of the signals was reduced from about 62 kHz to about 15,5kHz. An other problem was that only one channel could be frequency transformed at a time. The signals were not stable so means had to be taken to ensure that corresponding data from the signals were taken from the same starting points on the tape. This was done by using one track for sync pulses. Some of the results are shown in table 4,5 and 6. Each table give a number of measurements of the same signal.

The reduction in speed of the tape-recorder seemed to influence the phase of the signals. The frequency transformation also introduced errors as with the simulation with self generated signals. Some of the variations in the calculated values have come from the variations in the signals.

The results are therefore connected with some uncertainty. With this in mind the calculated values of σ are very near what was expected compared to the displayed directions on the DF-system's CRT. The correct directions of the incoming wave were not known, only the ellipses' directions could be seen on the CRT. The values of θ_1 can therefore not be verified.

Measure-results					Calculations		
$\theta_x-\theta_z$	$\theta_y-\theta_z$	A_x	A_y	A_z	θ_1	σ	T
77,9°	-39,2°	78,9	538	675,2	93,6°	93,9°	0,13
79,0°	-44,9°	76,5	529,5	668,5	93,8°	94,7°	0,12
105,9°	-35°	79,1	573,7	714,8	96,2°	96,2°	0,08
81,0°	-46,9°	73,2	499,7	633,9	94,2°	95,2°	0,11
117,6°	-51°	75	499,6	640,6	98°	98,4°	0,03
121,1°	-33°	64	440,9	565,3	97,8°	97,5°	0,06
136,2°	-42,7°	75,8	512	656,4	98,4°	98,4°	0,003
117,9°	-45,8°	69,9	531,4	667,6	96,9°	97,2°	0,04
85,9°	-47,1°	76,6	511,4	647,3	94,9°	95,9°	0,11
111,0°	-32,8°	76,1	498,2	614,9	97,3°	97,1°	0,09

Table 4

Measure-results					Calculations		
$\theta_x-\theta_z$	$\theta_y-\theta_z$	A_x	A_y	A_z	θ_1	σ	T
$-79,2^o$	$105,2^o$	15,8	32,5	30,2	$-65,8^o$	$-64,1^o$	0,03
$-66,8^o$	$98,2^o$	20,2	44,7	36,7	$-59,6^o$	$-66,2^o$	0,10
$-87,2^o$	$94,3^o$	29,2	53,7	43,7	$-62,3^o$	$-61,5^o$	0,01
-64^o	$98,4^o$	39,1	69,6	64,2	$-53,4^o$	$-61,3^o$	0,13
$-82,4^o$	100^o	25,7	46,1	31,9	$-62,4^o$	$-60,9^o$	0,02
$-40,3^o$	$97,2^o$	39,1	68,6	47,7	$-38,1^o$	$-64,4^o$	0,32
$-19,6^o$	$96,5^o$	28,8	49,8	33,3	$-28,9^o$	$-71,3^o$	0,48
$-65,5^o$	$85,7^o$	16,9	28,4	21,9	$-42,4^o$	$-60,9^o$	0,22
$-32,2^o$	$104,9^o$	36	62,9	49,8	$-43,4^o$	$-64,4^o$	0,32

Table 5

Measure-results					Calculations		
$\theta_x-\theta_z$	$\theta_y-\theta_z$	A_x	A_y	A_z	θ_1	σ	T
-153,3°	-33,2°	15,7	116,1	141,2	93,7°	93,9°	0,12
-136,8°	-30,9°	12,3	125,8	119,5	93,6°	91,5°	0,09
-135,6°	-43,3°	17,8	132,3	172,1	90,8°	90,3°	0,13
-173,5°	-44,5°	17,9	93,7	149,2	95,9°	97°	0,15
-161,9°	-46,5°	13,4	111	123,2	94,8°	93°	0,11
-149,3°	-30°	18,1	96,5	147	91,3°	95,4°	0,16
-155,5°	-35,5°	17,1	142,9	156,7	94,4°	93,5°	0,10
-125,5°	-34,2°	21,1	136,3	193,5	88,3°	90,2°	0,15
-169,9°	-41,2°	18,3	105,3	158,7	95,2°	96,3°	0,13
-130,1°	-43°	14,1	108,9	133,1	90,6°	89,6°	0,13
-75,8°	-41,3°	15,7	151	153,6	86,3°	85,1°	0,06

Table 6

4.2 Systems design

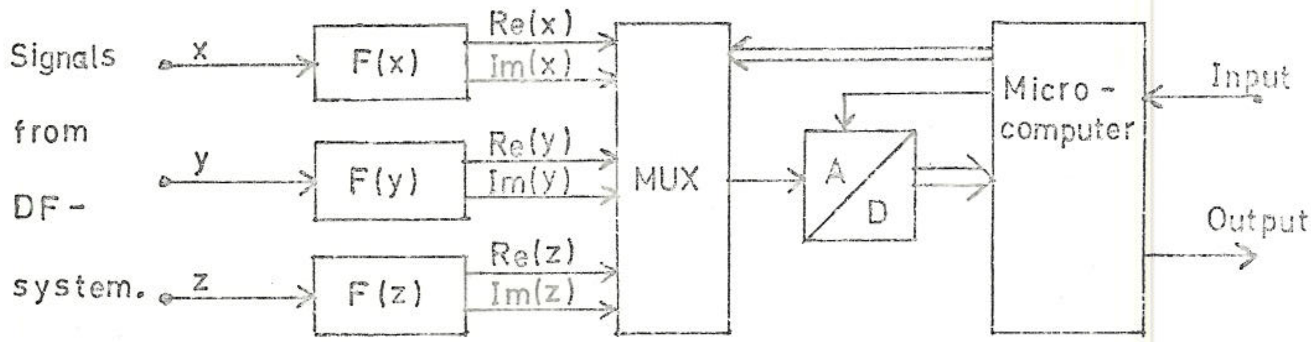

Figure 16 DF-extractor

Fig 16 shows a system to estimate the directions of signals from a DF-system. The three signals are fetched from the DF-system. Using a CZT-processor for each channel, the three signals are transformed into the frequency domain. The outputs are multiplexed, digitalized and fed into a microcomputer. In the computer the information from the frequency representation is extracted and the direction(s) are calculated.

The bandwidth of the signals from the DF-system is approximately 450 Hz (-3dB). When sampling the signal one must assume some larger bandwidth to be sure there is no signal outside the frequency band. The minimum sampling frequency of the CZT-processor is 4 kHz, which will cover a bandwidth of 2 kHz. This ought to be sufficient. With a bandwidth of 2 kHz divided between 256 frequency points, this will give a resolution of approximately 8 Hz. The output signal will consist of a real and imaginary part.

The A/D-converter will either have 8 bits or 12 bits resolution. The speed requirement is determined by the speed of the CZT-processor and the microcomputer. The speed of the processor is 4 kHz, which means a frequency component each 250 µsec. In 250 µsec six data have to be digitalized, which give 42 µsec pr data. In addition there is the time to fetch the data. It is estimated that the A/D-converter must have a conversion time of approximately 20 µsec with 12 bits resolution. A little slower with 8 bits resolution.

The microcomputer will be built with a 6800 microprocessor as the CPU.

4.3 Direction finding considerations

It has been shown how a system can be built that will calculate a direction or directions from signals from a DF-system. For a strong and continous signal the calculations will be straight forward and the result will be good.

Experience show that the received signals often are influenced by several factors:

- Bad signal/noise ratio
- Discontinuity in the incoming waves
- Variations in the signals parameters

These factors will mean that the calculated result will differ from calculation to calculation on the same signals, and means must be taken to reduce their influence.

Several noise sources are present in the system. Noise will come in on the antenna system from the surroundings. This can be regarded as equally distributed from all directions and white, which means that it is equally distributed in frequency. Noise will also be generated in the DF-system, distributed in frequency. A frequency representation will give a spectrum with noise over the whole frequency range while an incoming wave will be concentrated to a narrow band, and it is therefore easier to select out the seeked information and gaining a better signal/noise ratio.

Several kinds of signals are discontinuously, for instance morse and telegraphy. To find the direction of such a signal one must fetch information while the signal is on. In addition such signals will influence the AGC of the receiver.

A variation of the signals parameters might occur. For a wave with a reflection this will give an ellipse which vary in magnitude and direction. A method to reduce the influence of these error sources is to take a number of measurements, select acceptable data and calculate the direction for each of the datasets. From all those results a mean-value is calculated.

4.4 Time considerations

The time it takes to fetch one set of data is:

$$\frac{1}{4 \cdot 10^3 \text{Hz}} \cdot 512 = 0{,}128 \text{ sec}$$

When a number of data is to be used to calculate an average, the time for each dataset will be added to the total. For instance it will take 1,28 sec to fetch 10 datasets. In addition there is the dataselection and direction calculation, which might take at least 1 sec, but this is still to be investigated.

4.5 System operation

A whole DF-station will mainly consist of a DF-system, a direction extractor and a NORD-10 computer. The DF station will be remote controlled from a control station. The communication between the DF station and the control station will take place over a datanetwork to which the NORD-10 is connected. Fig 17 illustrates a total system-configuration.

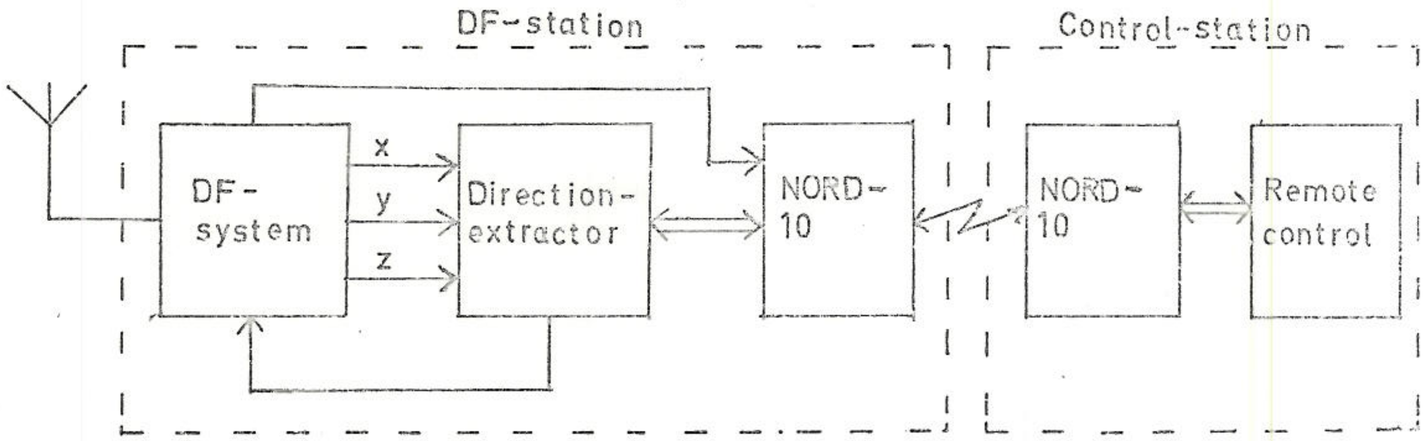

Fig 17 Remote control of a DF-station

Description of how it would function:

First the frequency range has to set on the DF-system. This is done from the remote control. Two 75 Hz channels are then opened between the DF station and the control station, one in each direction. Over one channel the operator in the control station is able to listen to what the DF station is tuned to. Two possibilities might then be available; automatic tuning of the direction finder or manually by the operator in order to obtain the best possible conditions. The manually tuning can be done over the other 75 Hz channel. When the tuning is done a commando "calculate the azimuth" is given, and when the DF station receives this commando it starts execution. The azimuth is calculated together with data about the goodness of the direction finding. The results are automatically transmitted back to the operator. If several azimuths are found the operator must be informed about this and be able to fetch them all. If the DF station fails to find any an error-message should be sent.

5 CONCLUSION

The theory show that a transformation of the signals to the frequency domain represent several advantages to our system:

- Separating signals with different frequencies
- Finding the signals amplitude and phase
- Obtain better signal/noise ratio

The signals are narrow-banded about 62 kHz. A simple method has been shown as how to obtain frequency zoom on that band just by a reduced sampling-rate. A CZT-processor is to be used to do the frequency transformation.

Simulation show that the principle of the system is very promising, but this remains to verify when a whole system has been built.

Two types of interference was discussed; coherent and incoherent. With the frequency representation the incoherent interference was no problem. For coherent interference two methods was used to estimate the direction:

- Using the ellipse's direction as the estimate
- Calculating the direction by using all information from the three channels

The first method is easiest to calculate and control. If the reflections are small the estimate will be good. The second method use both amplitude and phase of the third channel (z-signal). There is some uncertainty of this channel relative to the two others. In addition one must assume that there is only one reflection. The method is more difficult to control. For a start it is natural to select the first method, and this will be done.

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