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"On-line Determination of Apparent Resistivity in Magnetotelluric Soundings"

About three years ago we started work on a program of AMT soundings in the frequency range of 3 to 1000 Hz. Two years later the range was extended to the MT periods from 0.25 to about 3000 seconds. To increase the efficiency of our equipment we built it around a central processing unit, controlled by a 16 bit TM 9900 microprocessor. Thanks to the microprocessor we can achieve two aims. The first aim is to record a larger number of digital samples of the electric and magnetic field variations for future processing with big computers. The second aim is to process a sufficient selection of the data on the spot so as to allow the field crew to evaluate the sounding immediately: was the sounding successful, should the collection of sounding data be continued or are the man-made perturbations so strong that the sounding should be abandonned momentarily and rather be repeated later, perhaps at night when there are fewer perturbations. In effect what we are able to do right on the sounding spot are the following things:

1) Display the amplitude spectra of any of the five channels.

2) Display progressively improving apparent resistivity and phase data versus frequency on a scope screen.

Additional things, that are now routinely done in the laboratory with the very same equipment and which could also be carried out on the spot, are:

3) Rotation to the principal coordinate system if the data display 2-D characteristics.

4) If the data satisfy 1-D criteria, a 1-D inversion can be carried out.

Let us now describe the operation of our AMT system on the sounding site .

In the AMT range the soundings proceed in cycles of about 8 seconds duration. A cycle consists of a sampling stage, followed by stages of computing, displaying, and storing on magnetic tape. After some preliminary analog filtering, the two electric and three magnetic signals are amplified and low-pass filtered in two bands. The "high" band goes from about 1 to about 1000 Hz, the "low" band about from 1 to 150 Hz. The high band is sampled at the rate of 2 kHz until 2048 data points in each channel have been sampled and stored. The low band is sampled at the rate of 250 Hz until 256 sampling points have been acquired. Both samplings proceed simultaneously and are therefore terminated after 1.024 seconds.

The microprocessor then switches to a computing program. Both high and low samples are Fourier transformed by the Winograd FFT algorithm working on 240 points. The first 240 sampling points in each band are transformed and thus yield 120 complex Fourier transforms. Averages over six neighbouring frequencies of apparent resistivity ρ_a, phase φ, and coherency C are then computed, and when C exceeds or equals a pre-set level these average ρ_a and φ are dumped into bins of appropriate average frequency. There are 40 such bins for ρ_a and 40 for φ . Next the contents of each bin is divided by the number of samples admitted into the bin, which yields a kind of time-average, and this time-average is displayed on a scope screen, the ordinate being either $\log_{10}$ of ρ_a or φ and the abscissa being $\log_{10} f$ or $\log_{10} T$. Finally, the entire contents of the memory is transferred to a digital magnetic tape.

This constitutes a complete cycle of our AMT soundings. The cycle repeats every eight seconds, unless the signal in any of the channels saturates the corresponding amplifier during the one second sampling stage. When this happens the cycle

FLOWCHART OF AMT-SOUNDING MICROCOMPUTER PROGRAM

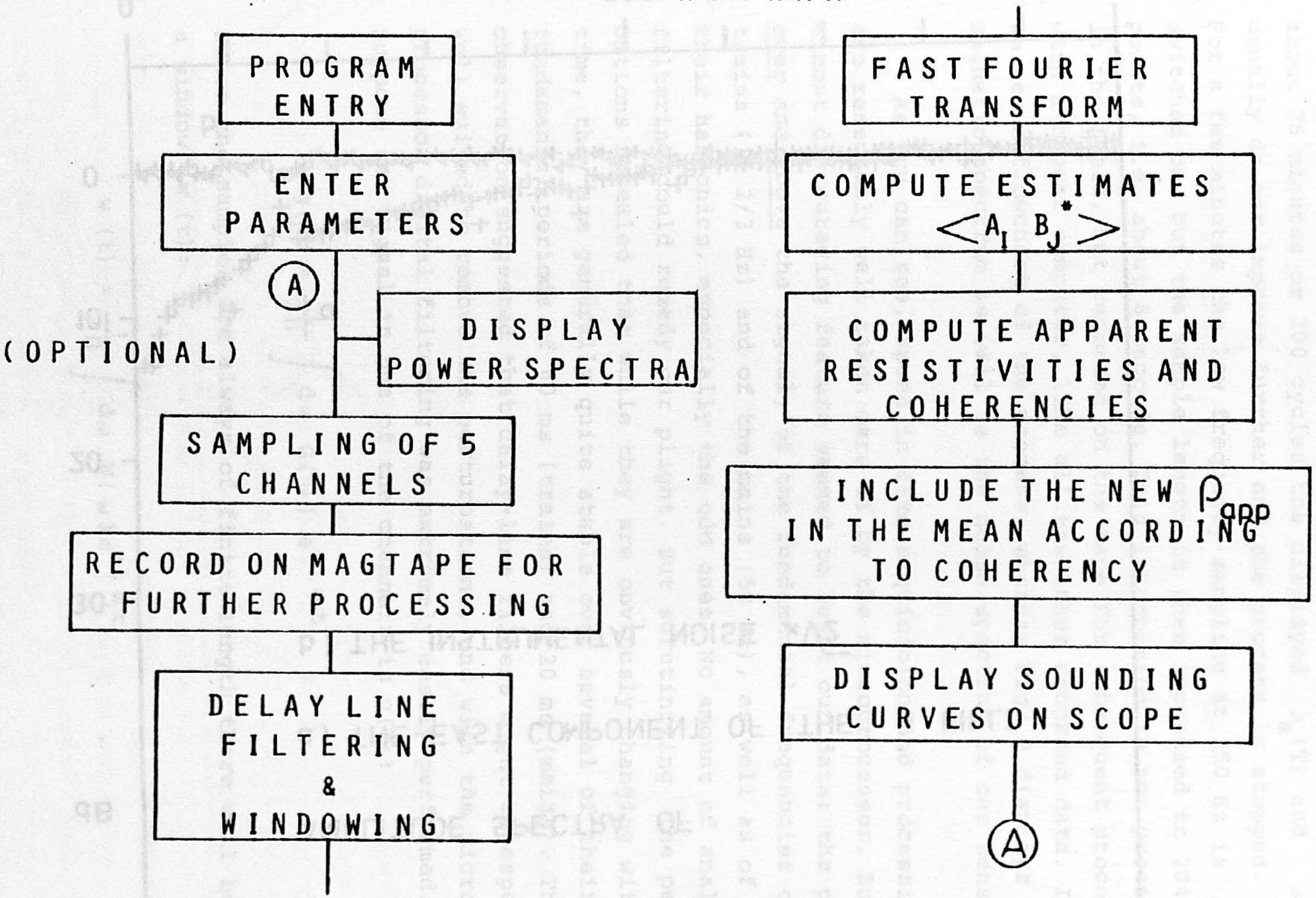


Fig. 1

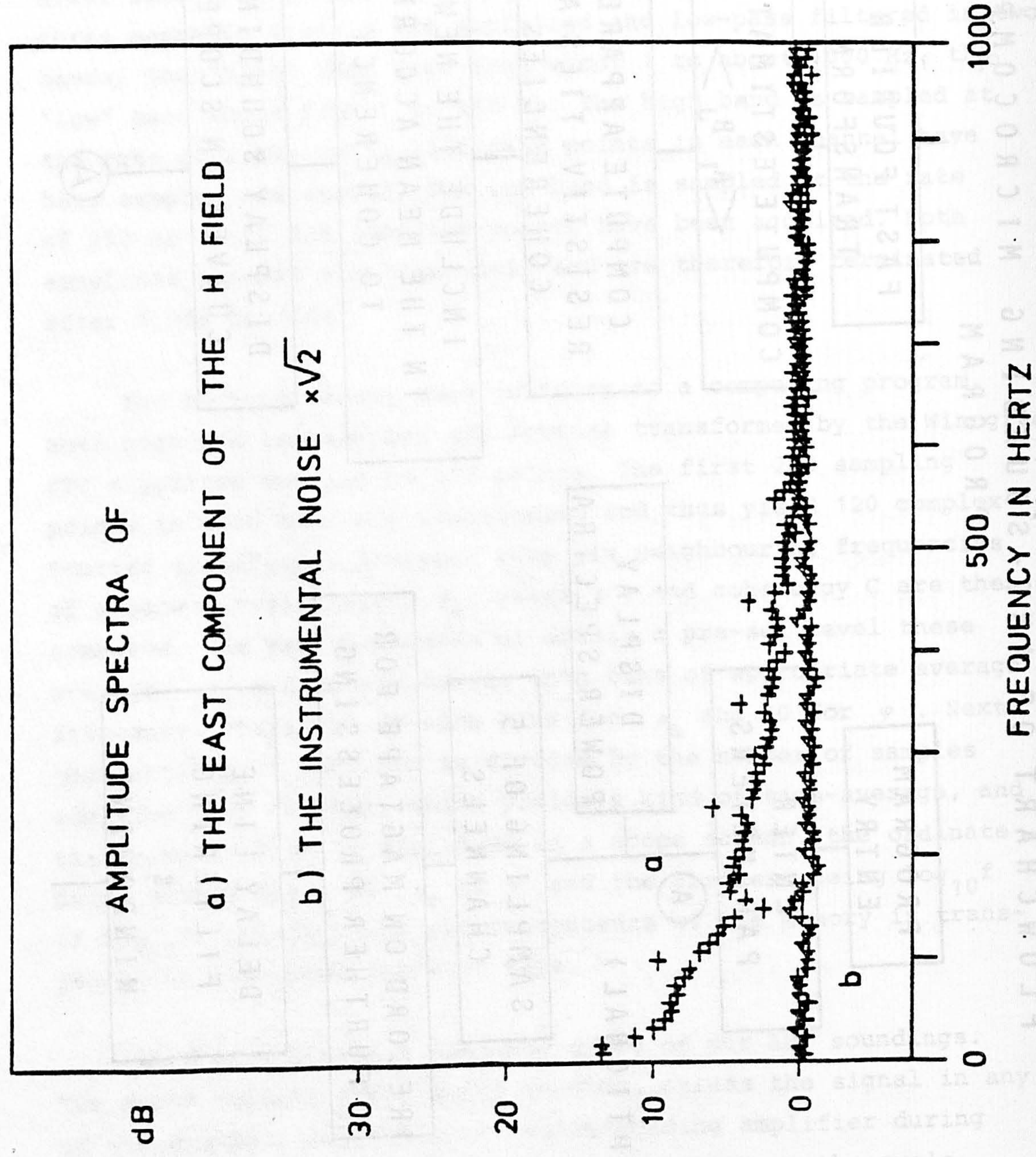

AMPLITUDE SPECTRA OF
a) THE EAST COMPONENT OF THE H FIELD
b) THE INSTRUMENTAL NOISE ×√2
dB
30
20
10
0
0
500
1000
FREQUENCY IN HERTZ
a
b

Fig. 2

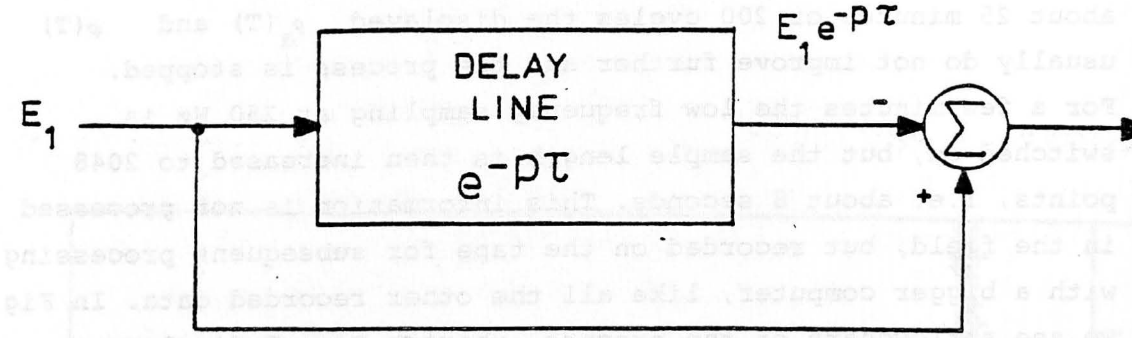
starts anew. As time proceeds the various bins gradually fill up with more estimates of ρ_a and φ, so that the scatter of the $\rho_a(T)$ and $\varphi(T)$ data points continually decreases. After about 25 minutes or 200 cycles the displayed $\rho_a(T)$ and $\varphi(T)$ usually do not improve further and the process is stopped. For a few minutes the low frequency sampling at 250 Hz is switched on, but the sample length is then increased to 2048 points, i.e. about 8 seconds. This information is not processed in the field, but recorded on the tape for subsequent processing with a bigger computer, like all the other recorded data. In Fig. 1 we see a flowchart of the process, whereas Fig. 2 displays a magnetic spectrum as well as the noise spectrum of our sensors.

As you can see, speed in data acquisition and processing are reasonably well taken care of by the microprocessor. But a most discouraging feature seemed to beset our data: the presence, over and above the signal, of the fundamental frequencies of the trains (16 2/3 Hz) and of the mains (50 Hz), as well as of all their harmonics, especially the odd ones. No amount of analog filtering could remedy our plight. But scrutinizing the perturbations revealed that while they are obviously changing with time, they are generally quite stable over several of their fundamental periods of 60 ms (trains) and 20 ms (mains). This observation suggested that delay-line filters might be especially well suited to remove the perturbations, and with the microprocessor digital filtering was extremely easily performed. Suppose the signal in one of the channels is g(t):

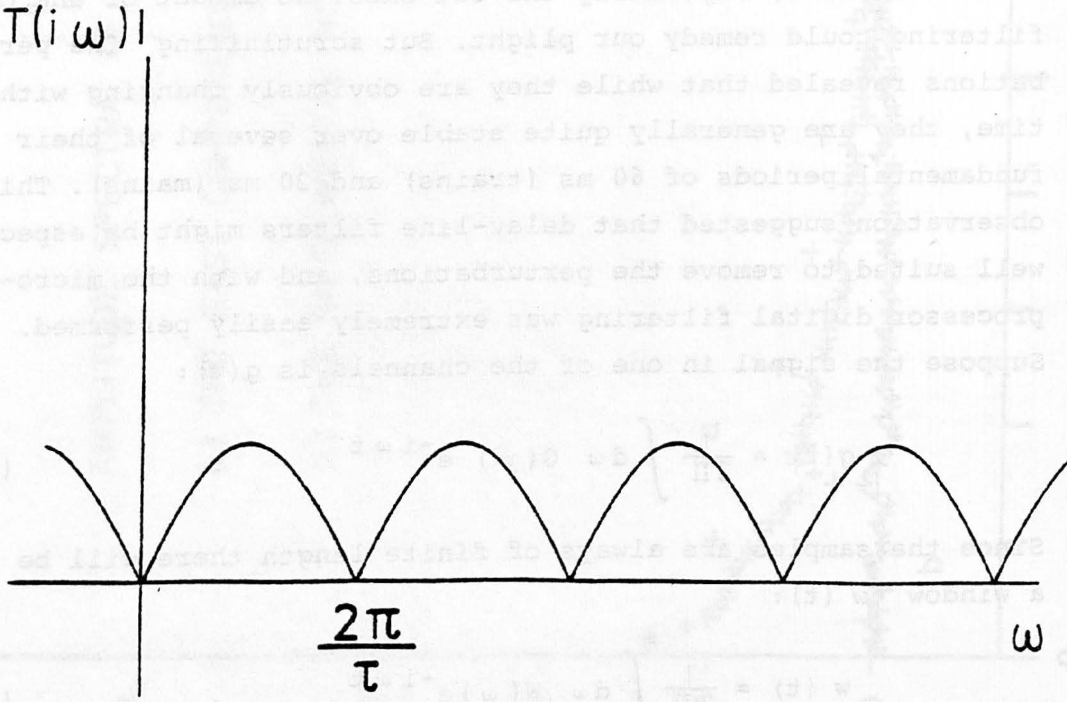
$$g(t) = \frac{1}{2\Pi}\int d\omega \; G(\omega)\, e^{-i\omega t} \quad . \qquad (1)$$

Since the samples are always of finite length there will be a window w (t):

$$w(t) = \frac{1}{2\Pi}\int d\omega \; W(\omega)\, e^{-i\omega t} \quad , \qquad (2)$$



a) BLOCK DIAGRAM OF A CANCELLER



b) FREQUENCY RESPONSE

Fig. 3

so that the sampled signal is effectively w(t).g(t), and as you know, the spectrum of this product is a convolution of the two spectra $G(\omega)$ and $W(\omega)$. Suppose, however, that we use a delay-line filter, such that the filtered signal we store and process becomes

$$w(t)\ [\ g(t) - g(t-\tau)\ ]\quad . \tag{3}$$

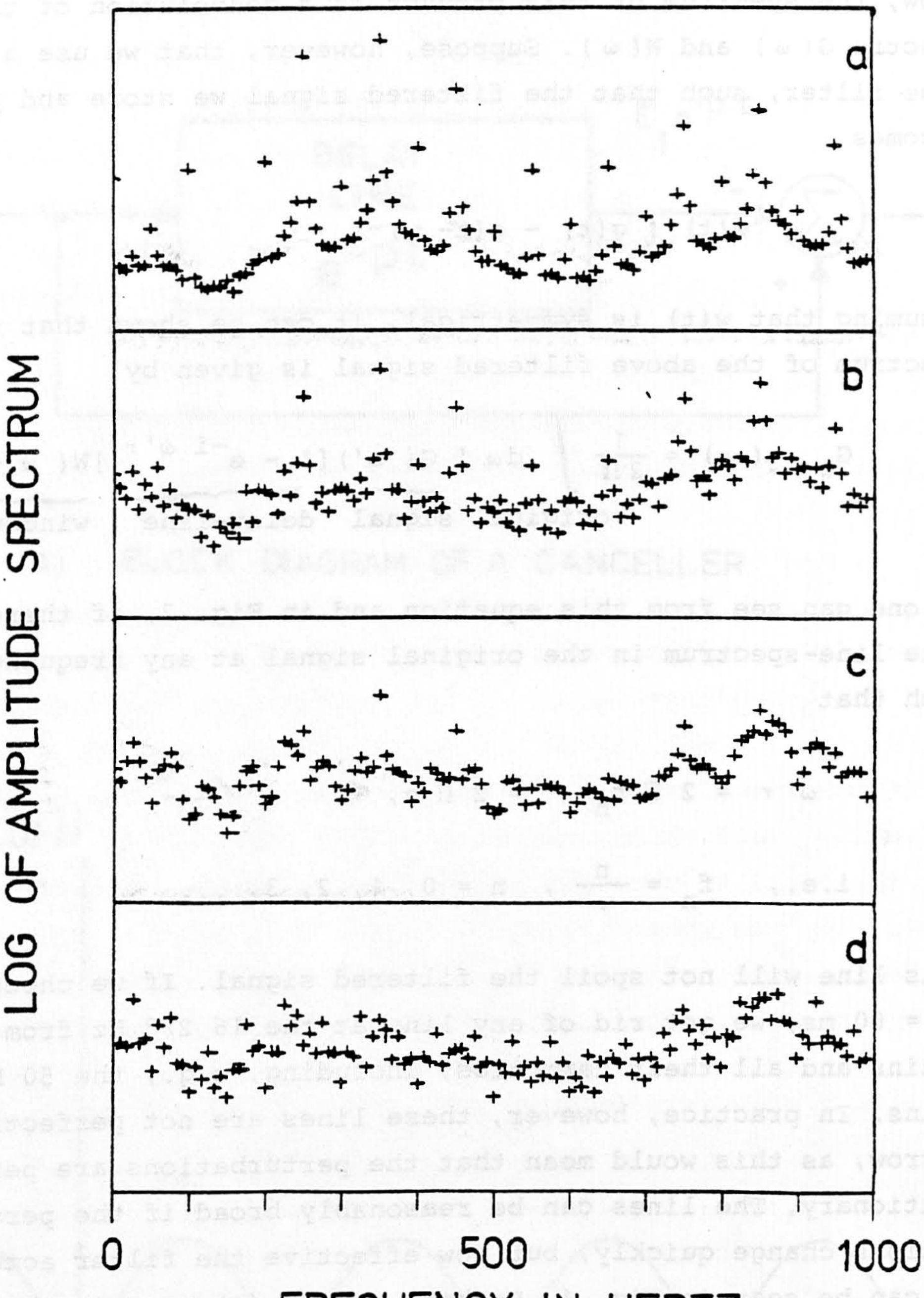
Assuming that w(t) is symmetrical, it can be shown that the spectrum of the above filtered signal is given by

$$G_{W\tau-}(\omega) = \frac{1}{2\Pi}\int d\omega'\ \underbrace{G(\omega')}_{\text{original signal}}\underbrace{[1 - e^{-i\,\omega'\tau}]}_{\text{delay-line}}\underbrace{W(\omega'-\omega)}_{\text{window}} . \tag{4}$$

As one can see from this equation and in Fig. 3, if there is a true line-spectrum in the original signal at any frequency f_n such that

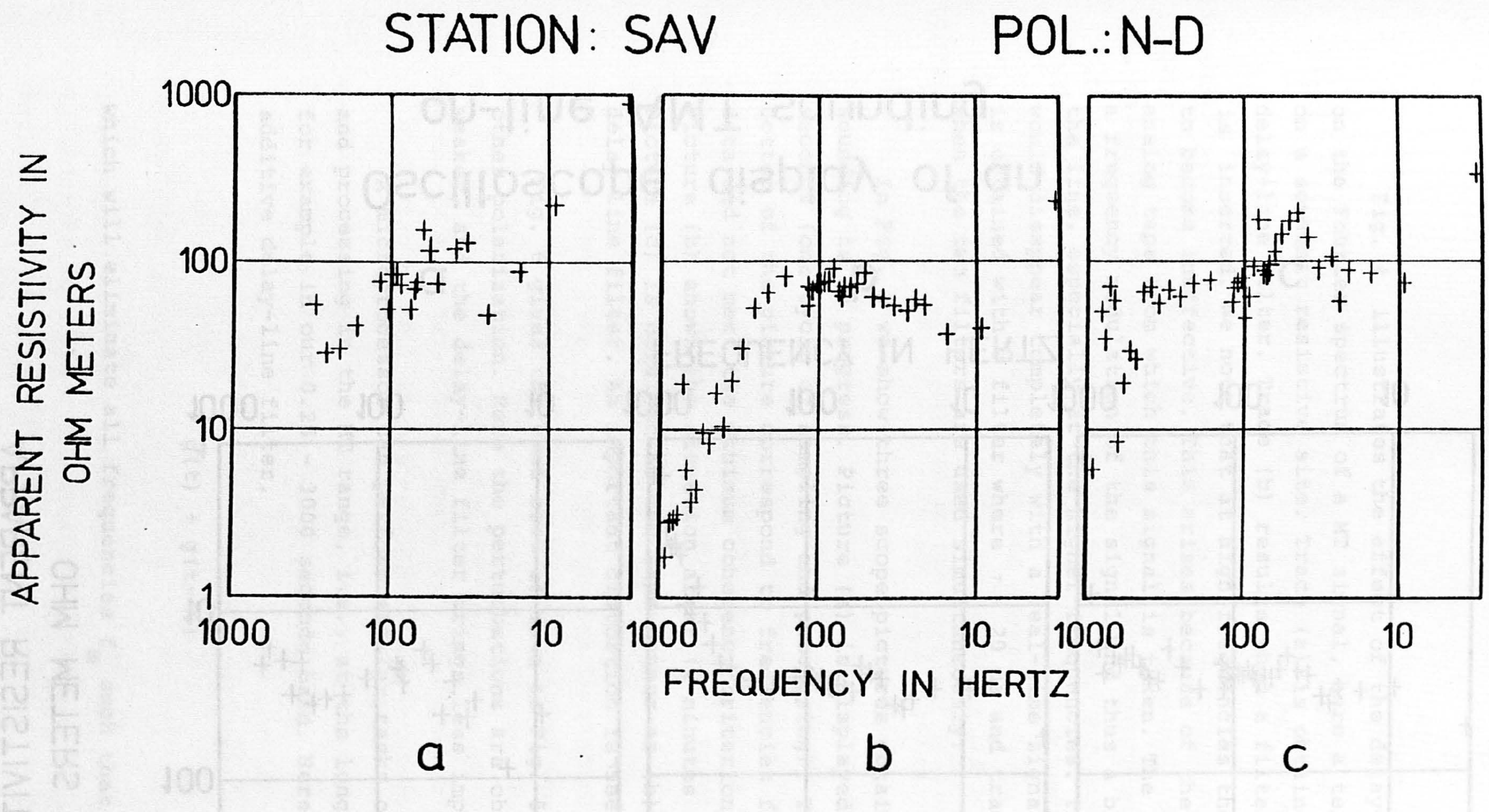
$$\omega'\tau = 2\,\Pi f_n\tau = 2\,\Pi n,$$
$$\text{i.e.,}\quad f_n = \frac{n}{\tau}\ ,\quad n = 0, 1, 2, 3, \ldots\quad , \tag{5}$$

this line will not spoil the filtered signal. If we choose τ = 60 ms, we get rid of any line at the 16 2/3 Hz from the trains and all their harmonics, including, e.g., the 50 Hz mains. In practice, however, these lines are not perfectly narrow, as this would mean that the perturbations are perfectly stationary. The lines can be reasonably broad if the perturbations change quickly, but how effective the filter actually is can be seen in Fig. 4. Looking at Eq. (4) we note that among the frequencies rejected from $G(\omega')$ we find the very low frequencies. The subtraction filter described by Eq. (3) is therefore also a "high-pass" filter. Furthermore, it can be shown that the power contained in the filtered signal is about twice the white noise power of the original signal, implying that g(t) and $g(t-\tau)$ are uncorrelated.



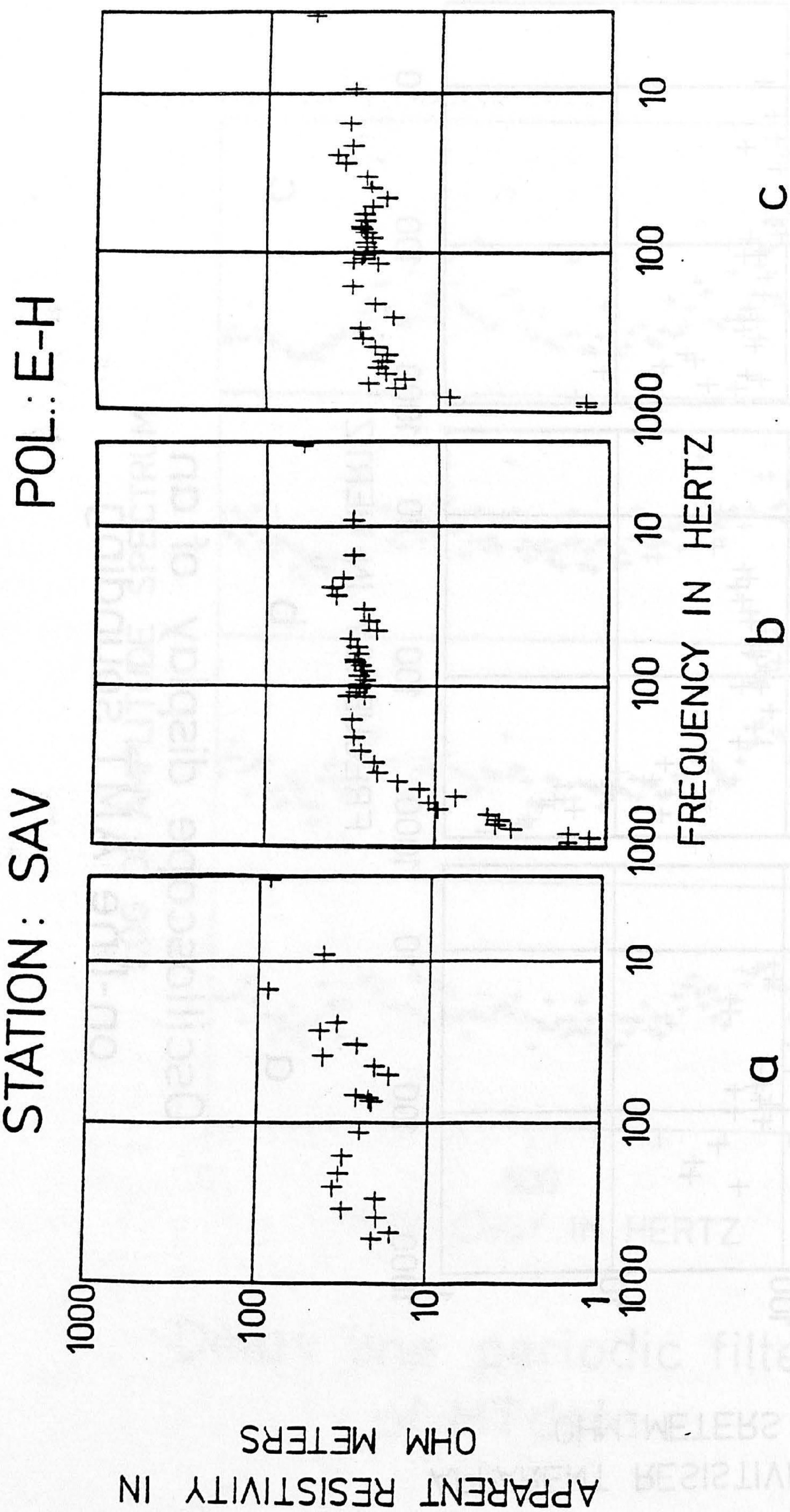
Delay line periodic filtering of MT data

Fig. 4



Oscilloscope display of an on-line AMT sounding

Fig. 5



Oscilloscope display of an on-line AMT sounding

Fig. 6

Fig. 4 illustrates the effect of the delay-line filters on the Fourier spectrum of a MT signal, here a telluric component on a somewhat resistive site. Trace (a) is obtained with no delay-line filter. Trace (b) results when a filter with $\tau = 60$ ms is inserted. We note that at high frequencies the filter appears to become ineffective. This arises because of the flutter of the analog tape from which this signal is taken. The flutter produces a frequency modulation of the signal and thus a broadening of the line, especially at the higher frequencies. This effect would disappear completely with a real-time signal. Trace (c) is obtained with a filter where $\tau = 20$ ms and trace (d) arises when the two filters are used simultaneously.

In Fig. 5 we show three scope pictures obtained while a sounding is in progress. Picture (a) is displayed after 8 seconds (one cycle of sampling and processing). The dots at the bottom of the picture correspond to frequencies for which the data did not meet the minimum coherency criterion imposed. Picture (b) shows the situation after 12 minutes (90 cycles). Picture (c) is obtained in the same manner as (b), but without delay-line filter. An important distortion is seen around 50 Hz.

Fig. 6 gives the same sort of data as Fig. 5 but for the other polarization. Here the perturbations are obviously much weaker, and the delay-line filter brings less improvement.

A microprocessor can perform similar tasks of data handling and processing in the MT range, i.e., at the long periods, as for example in our 0.25 - 3000 seconds band. Here we use an additive delay-line filter,

$$g(t) + g(t+\tau) \qquad (6)$$

which will eliminate all frequencies f_m such that

$$f_m = \frac{2m+1}{2\tau} \quad , \quad m = 0,\ 1,\ 2,\ \ldots \qquad . \qquad (7)$$

Obviously we have here a "low-pass" filter which can eliminate a fundamental frequency and all its odd harmonics.

In conclusion it can be said that the use of a microprocessor can greatly increase the flexibility of MT sounding equipment, without noticeably increasing its weight or power requirement. This is true not only at the high AMT frequencies, but also at much lower frequencies. The most important gain is the dialog it permits between the operator and the measurement. In addition all problems of automation, such as monitoring of signal level or quality, can be handled with very simple logistic means.