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ACOUSTICAL IMPEDANCE MEASUREMENTS BY THE TWO-MICROPHONE-THREE-CALIBRATION (TMTC) METHOD

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Abstract: We discuss a method for measuring the acoustical impedance of wood-wind instruments which is relatively fast and insensitive to the ambient acoustical noise. The apparatus is essentially made of a measurement head, which is an approximately cylindrical cavity fitted with two microphones (in practice often three or more for technical reasons) and, at one of its ends, a loudspeaker; the other end is connected to the acoustical cavity under study (the instrument). The loudspeaker emits acoustical chirps and the signals provided by the microphones are Fourier transformed with the help of a microcomputer, which also extracts from the results the acoustical impedance as a function of frequency. A relatively high dynamic range is needed in order to match the high quality factor of the resonances of musical instruments, especially at low frequencies. To meet this requirement, it was found necessary to take into account the acoustical losses inside the head as well as the mutual perturbation of the microphones or various perturbations arising from geometrical imperfections of the cavity. This can be done automatically and without long calculations by using a calibration method, based on the successive use of three reference cavities, which we describe and discuss in detail in this article. The method has been tested with cylindrical cavities and gives satisfactory results in terms of peak positions and heights. Impedance curves of clarinets have also been measured. The three-calibration technique eliminates several sources of systematics and corrects automatically for various perturbations, such as perturbations of the acoustical field by the microphones, and allows to study instruments of various diameters.

1. INTRODUCTION

It is well known that, for understanding the physical functioning of wood-wind instruments, a crucial quantity to consider is the input acoustical impedance [1] [2]. Various methods have been elaborated in the laboratory to measure the impedance of music instruments (see for example [3] [4]). Although most of them are perfectly satisfactory as research tools for physicists, they are not so easy to use for non-scientists and often require the sound shielding of an anechoic chamber to eliminate the ambient acoustical noise which would otherwise perturb the measurement. Moreover, the time taken by a measurement is often long, especially if the quality factor of the acoustical resonances of the instrument is high.

On the other hand, because impedance measurements give an objective characterization of woodwinds instruments, totally free of irreproducibilities introduced for instance by a reed or the player himself, one may hope to see one day that these measurements have become a normal control procedure in the workshop of instrument makers. To reach this aim, it is nevertheless clear that faster and more convenient impedance bridges are needed. This is why we decided, some time ago, to improve the existing methods so they

could provide quick acoustical impedance measurements which could be implemented more easily in a normal (but quiet) acoustical environment, while reproducing correctly the fast variations due to sharp resonances. The aim of the present article is to report the results that we have obtained while fulfilling this program.

Acoustical impedances are defined as the ratio between pressures, which are easily measured by microphones, and acoustical flows which are much more difficult to measure directly (although hot wire techniques have been used successfully for this purpose [5]). This is why, almost always, an indirect method is used to measure the flow, which is actually known through a pressure measurement. The most popular method is probably that which requires the use of a narrow capillary of high acoustical impedance put in series with the unknown impedance (exactly as, in electricity, high impedances in series are used to produce known currents, which are calibrated through the measurement of a voltage across a resistor). There is nevertheless a problem with this method: only a very weak fraction of the acoustical vibration amplitude which is available at the first microphone can go through the capillary tube, so that the second microphone is always restricted to a very small signal, which is easily perturbed by the ambient acoustical noise. Except in extremely quiet environments, expensive sound insulation is consequently needed.

In another class of experiments, the narrow capillary is replaced by a broader tube, which has actually an internal diameter of the same order of magnitude than the instrument itself; in this "measurement head", much more intense acoustical signals are available at two microphones which are fitted along the tube. Of course, it is no longer possible as in the capillary method to consider that one single microphone measures the pressure, the other the flow: both physical quantities are actually mixed in the two signals. Nevertheless, with help of a microcomputer they can be sorted out so that, eventually, the acoustical impedance can be measured.

The general idea behind this class of methods is, of course, not new: it is actually strongly reminiscent of the method of the Kundt tube [6] and the basic theory can be found in textbooks on acoustics [7]; it has been used by numerous authors ([8] to [19]) for various purposes, ranging from characterization of the vocal tract to impedance measurements. Nevertheless, in the corresponding publications, the losses inside the measurement head are not always carefully taken into account (they are often merely ignored) and no attempt is made to obtain the high dynamic range which is necessary to measure the narrow resonance peaks of acoustical resonators, such as closed cylinders or real musical instruments. For this purpose, and for eliminating various perturbations such as geometrical imperfections of the measurement head, non-zero admittance of the microphones, etc. we had to elaborate a calibration method which provides the relation between the pressure measured by the microphones and the acoustical pressure and flow at the reference plane (the plane where the impedance is measured). In other words, we obtain this relation from auxiliary experiments and not only from theory. This is the major difference between our work and that for example of ref. [12] to [14] (our theoretical analysis, based solely on the continuation of solutions of the wave equation, is also somewhat different). Our method requires the use of three auxiliary cavities, in a way which automatically eliminates several sources of systematics and allows precise measurements of high impedance peaks; its general principle is described in the next section.

2. THE METHOD

2.1 GENERAL IDEA

Figure 1 shows a sketch of the general set-up that we study here: the impedance to be measured is connected to a measurement head, which may be cylindrical or have another geometrical shape, and which is fitted with two microphones; at its other end, the measurement head has the shape of a closed cavity containing a loudspeaker which provides the acoustical excitation. We first assume that the excitation is monochromatic and study the response of the system at a single frequency. We also assume that the neck of the acoustical cavity to be studied is almost cylindrical and that, inside this neck, the acoustical wave contains only one single transverse mode; in other words, in this region, the propagation of the acoustical wave is equivalent to a one-dimensional problem, so that the solution of the wave equation can be obtained (in theory) from the value of the acoustical field and its space derivative at one point (as in quantum mechanics for example, where the solution of the Schrödinger equation in a one dimensional problem can be obtained from the value of the wave function and its derivative at one point). We can then define the wave by two parameters, the values p of the pressure and u of the acoustical velocity in a reference plane that we choose at the end of the neck (the plane where we wish to measure the impedance). By continuation of the solution towards the inside of the measurement head, one can obtain the acoustical wave everywhere inside the head, and in particular write the signals s_1 and s_2 provided by the two microphones as linear functions of p and u :

$$\begin{cases} s_1 = \alpha p + \beta \rho c u \\ s_2 = \gamma p + \delta \rho c u \end{cases} \quad (1)$$

where α , β , γ and δ are unknown parameters which depend on the geometry of the system, as well as on the gain of the microphones; ρ is the mass of gas (air) per unit volume and c is the velocity of sound (these factors have been introduced for pure convenience, for all unknown parameters to have the same dimension). Now, if the wave obeys the boundary conditions which are required in the definition of the acoustical impedance, that is if it is a purely outgoing wave outside the open end of the cavity under study (this is indeed the case if the sound is produced by the loudspeaker only and emitted through the bell or the open holes of the instrument), one has:

$$Z = \frac{p}{u} \quad (2)$$

where Z is the impedance of the cavity to be measured. Consequently, if one defines y as the ratio of the two signals provided by the microphones, one has:

$$y = \frac{s_2}{s_1} = \frac{\gamma Z + \delta \rho c}{\alpha Z + \beta \rho c} \quad (3)$$

or:

$$Z = \rho c \frac{-\beta y + \delta}{\alpha y - \gamma} = \rho c \frac{Ay + B}{y - y_0} \quad (4)$$

with:

$$\begin{aligned} A &= -\beta/\alpha \\ B &= \delta/\alpha \\ y_0 &= \gamma/\alpha \end{aligned} \quad (5)$$

The problem now is to determine the values of the three unknown coefficients A , B and y_0 . This can be done by steps with the help of three auxiliary calibration measurements where the unknown impedance is removed and replaced by other systems of known acoustical characteristics. First, a system with zero admittance (infinite impedance) in the plane of reference is substituted to the cavity under study, which is easily done by simply closing the head by a rigid surface in the reference plane (the admittance in this plane is not strictly zero because of the isothermal boundary conditions on a solid plane [20] [21], but the corresponding correction is negligible: the thermal perturbation extends in the air over a length which is inversely proportional to the frequency, and of the order of a fraction of a millimeter at 100 Hz.); then, the ratio y is measured, which immediately provides the value of y_0 [see expression (4)]. Second, two reference cavities with known acoustical impedances Z' and Z'' are used in succession and the corresponding values y' and y'' are measured; from (4) one gets the following relations:

$$\begin{aligned} \bar{Z}' (y' - y_0) &= A y' + B \\ \bar{Z}'' (y'' - y_0) &= A y'' + B \end{aligned} \quad (6)$$

(where \bar{Z}' and \bar{Z}'' are the dimensionless impedances, i.e. normal impedances divided by ρc), a system of two linear equations in A and B which has the solutions:

$$\begin{aligned} A &= \frac{\bar{Z}' (y' - y_0) - \bar{Z}'' (y'' - y_0)}{y' - y''} \\ B &= \frac{\bar{Z}'' y' (y'' - y_0) - \bar{Z}' y'' (y' - y_0)}{y' - y''} \end{aligned} \quad (7)$$

Using these equations, the values of A and B can be obtained from a measurement of y' and y'' as soon as impedances \bar{Z}' and \bar{Z}'' are known; then, relation (5) can be used to derive the value of the unknown impedance Z from a measurement of y according to:

$$Z = \rho c \frac{\bar{Z}' (y' - y_0)(y - y'') + \bar{Z}'' (y'' - y_0)(y' - y)}{(y - y_0)(y' - y'')} \quad (8)$$

As this formula shows, what is really done by this method is to compare the unknown impedance with three reference impedances (including one which is infinite; see the remark at the end of this section for a generalization of (8) to the case where all impedances are finite).

In practice, the acoustical wave generated by the loudspeaker is not a pure sinusoid, but rather a "chirp" containing a whole range of frequencies, or sometimes a series of chirps (see §3). After storing the time signals provided under these conditions by the two

microphones, a microcomputer Fourier transforms both signals and, for each frequency of interest $f = \omega/2\pi$, calculates the ratio $y(\omega)$. Finally, formula (8) is used to obtain the acoustical impedance $Z(\omega)$.

By this procedure, several sources for systematic errors are automatically eliminated:

* since the only relevant combination of measured quantities is the ratio s_1/s_2 , no reproducibility of the acoustical chirps is needed. What does really matter is that the coefficients A , B and y_0 remain constant; in other words, the central part of the measurement head has to keep exactly the same acoustical characteristics during one given set of measurements, including the three calibrations.

* similarly, the two microphones do not have to be identical, and in particular their linear (complex) gain may be different; this because expression (8) is a homogeneous expression of degree zero with respect to the y 's, so that multiplying all the y 's by a complex coefficient does not affect the value of Z (even if this coefficient is frequency dependent). What is needed is only a good reproducibility in time of the characteristics of each microphone.

* Along the same line, one may remark that the exact way that the microphones are coupled to the acoustical field inside the head is not essential; if they are for example coupled through an acoustical duct, a narrow channel, or/and a cavity which change the phase and the amplitude of the pressure, the perturbation is completely eliminated in expression (8), provided of course that the coupling remains exactly the same during the complete set of measurements.

* On the other hand, we have always assumed that the acoustical waves propagate inside all the apparatus in a purely linear regime. It is clear that non-linear effects such as turbulence would introduce errors in the measurement; it is therefore necessary to ensure that this is not the case by performing measurements at different excitation levels, and checking carefully that the same values of Z are obtained.

* Also, we have assumed that all acoustical signals are due only to the vibration of the air column inside the impedance head, and that they propagate from the loudspeaker towards the measured impedance. If spurious signals due for example to the mechanical vibrations of the walls were detected by the microphones, or if acoustical signals were propagating from the outside of the experiment to the inside of the measured impedance and then to the microphones, the measurements could be strongly perturbed and incorrect results obtained.

Remark: The reasoning which leads to (8) can easily be generalized to the case where all calibration impedances are finite; one then obtains:

$$Z = \rho c \frac{\bar{Z}'\bar{Z}''''(y' - y'''')(y - y'') + \bar{Z}'\bar{Z}''(y'' - y')(y - y''') + \bar{Z}''\bar{Z}''''(y'''' - y''')(y - y')}{\bar{Z}''''(y'' - y')(y'''' - y) + \bar{Z}'(y'''' - y''')(y' - y) + \bar{Z}''(y' - y'''')(y'' - y)} \quad (8')$$

which generalizes equation (8) and reduces to it when \bar{Z}'''' is infinite and y'''' replaced by y_0 . It is easy to check on (8') that one obtains the value Z_i for the impedance when $y = y_i$.

2.2 A SPECIAL CASE: PERFECTLY CYLINDRICAL HEAD

We now study the case where the acoustical wave propagates inside a perfectly cylindrical duct of constant diameter in both the measurement head and the neck of the impedance under study. This more restrictive assumption will allow us in this section to obtain explicit expressions for the calibration functions A , B , and y_0 , which depend on the position of the two microphones, at distances L_1 and L_2 from the reference plane.

Considering a monochromatic wave inside a cylinder, and assuming that only the lowest transverse mode (piston mode) contributes, one can use simple considerations based on translation invariance to write the following expression for the pressure p and the acoustical velocity u (averaged over the transverse coordinates y and z):

$$\begin{cases} u = -i\omega (\mu_+ e^{ikx} + \mu_- e^{-ikx}) \\ p = \frac{-ik}{\chi} (\mu_+ e^{ikx} - \mu_- e^{-ikx}) \end{cases} \quad (9)$$

where x is the space coordinate along the axis of the cylinder, k the wave number, and χ the compressibility of the gas.

In the absence of dissipation, the sound velocity c is related to χ and ρ by the equality:

$$\chi\rho = c^{-2} \quad (10)$$

and k is merely equal to ω/c . The effects of dissipation (viscosity, heat conduction) change the value of k , which is now given by the equation [22] [23] [24]:

$$k^2 = \frac{\omega^2}{c^2} \left[1 + (1-i) \frac{\sqrt{2}}{R} \epsilon_{vt} \sqrt{\frac{c}{\omega}} \right] \quad (11)$$

where R is the radius of the cylinder and where ϵ_{vt} is a quantity, homogeneous to the square root of a length, which characterizes the dissipation in the gas; we refer to the above references for a detailed expression of ϵ_{vt} as a function of the heat conductivity and the viscosity coefficient of the gas. Expression (11) is obtained by assuming isothermal conditions on the walls of the cylinder. It is valid only to first order in ϵ_{vt} and assumes that R is much larger than the thickness of the boundary layer; the conditions are easily satisfied when the wavelength is that of audio waves propagating inside cylinders with a diameter of the order of one centimeter.

From (9), we can now calculate the ratio y of the signals provided by the two microphones at distances L_1 and L_2 from the reference plane (here we completely neglect any perturbation of the plane acoustical wave created by the presence of the microphones) and obtain:

$$y = \frac{p_2}{p_1} = \frac{\mu_+ e^{ikL_2} - \mu_- e^{-ikL_2}}{\mu_+ e^{ikL_1} - \mu_- e^{-ikL_1}} \quad (12)$$

Equation (9) can also be used to calculate the acoustical impedance Z in the plane of reference ($x = 0$), and to write:

$$Z = \frac{p(x=0)}{u(x=0)} = \frac{k}{\chi\omega} \frac{\mu_+ - \mu_-}{\mu_+ + \mu_-} = \rho c \left(\frac{ck}{\omega} \right) \frac{(\mu_+/\mu_-) - 1}{(\mu_+/\mu_-) + 1} \quad (13)$$

But relation (12) can easily be inverted to give the ratio μ_+/μ_- as a function of y , so that one finally obtains:

$$Z = \rho c \left(\frac{ck}{\omega} \right) \frac{y (e^{-ikL_1} - e^{ikL_1}) - (e^{-ikL_2} - e^{ikL_2})}{y(e^{-ikL_1} + e^{ikL_1}) - (e^{-ikL_2} + e^{ikL_2})} \quad (14)$$

When k is complex and contains a small imaginary part δk_i :

$$k = k_r + i \delta k_i \quad (15)$$

on has:

$$\cos(kL) = \cos k_r L \cosh \delta k_i L - i \sin k_r L \sinh \delta k_i L \quad (16 - a)$$

and:

$$\sin(kL) = \sin k_r L \cosh \delta k_i L + i \cos k_r L \sinh \delta k_i L \quad (16 - b)$$

and (14) can be rewritten in the form:

$$Z(\omega) = -i\rho c \left(\frac{ck}{\omega} \right) \frac{y \sin(kL_1) - \sin(kL_2)}{y \cos(kL_1) - \cos(kL_2)} \quad (17)$$

As expected, this expression [as well as (14)] has indeed the general form (4), with the following values for the coefficients A , B and y_0 :

$$A = -i \left(\frac{ck}{\omega} \right) \frac{\sin(kL_1)}{\cos(kL_1)} \quad (18 - a)$$

$$B = i \left(\frac{ck}{\omega} \right) \frac{\sin(kL_2)}{\cos(kL_1)} \quad (18 - b)$$

$$y_0 = \frac{\cos(kL_2)}{\cos(kL_1)} \quad (18 - c)$$

We emphasise again that these relations are only approximate, if only because they assume that both microphones have zero admittance and do not perturb in any way the sound wave inside the cylinder. If they were used, our measurements would become equivalent to those that can be found in the literature ([8] to [19]); here, they will be used only as guides and as points of comparison.

3. EXPERIMENTAL SETUP

Figure 2 shows the general features of the experimental arrangement which was designed to perform acoustical impedance measurements with the TMTC method, based on the use of formula (8). The measurement head is made of brass, with a sufficient thickness (7 millimeters) to reduce mechanical vibrations which otherwise would perturb the measurement. The loudspeaker is a medium range dome type, 12 cm. in diameter, and it is mounted inside a box with a conical tube to ensure a better coupling towards the cylindrical part of the head, 15 millimeters in diameter (the bore of a standard clarinet).

Otherwise, this box is completely closed, so that no backward radiation is emitted. It is desirable to avoid strong resonances inside the corresponding cavity, because they would introduce an unnecessary strong frequency dependence of the acoustical level detected by the two microphones, which would result in severe dynamics requirements on the measurements; for this purpose, it was found useful in practice to insert a piece of synthetic foam between the loudspeaker and the microphones, but of course this has to be done with moderation in order to avoid too much loss of sound intensity. The loudspeaker is excited by a 20 watt power amplifier, which itself is excited by an electrical chirp generated by the microcomputer (see below).

Although in principle two microphones are sufficient, in practice it is useful (if not indispensable!) to have more, three or even four. We now discuss why this is the case. Physically, it is obvious that, if the microphones are too close to each other, namely much less than one wavelength apart, the two signals y_1 and y_2 are not independent; instead, they can only take on values which are almost the same, whatever measurement is done, so that no real information is gained by measuring their ratio. Moreover, for a cylindrical head, this unfavourable situation occurs, not only if the distance $L_1 - L_2$ is almost zero, but also if it is an integer multiple of the acoustical wavelength λ (or actually even of half this wavelength): one then always obtains either equal (or opposite) values for y_1 and y_2 , and again no information is gained in the measurement of their ratio.

Mathematically one can see that, in formula (8), when all values of the y 's are equal, both the numerator and the denominator vanish so that the expression which gives Z becomes indeterminate. When the values are only slightly different, the determination of Z remains theoretically possible, but is in practice very inaccurate: it requires the knowledge of the difference between various y 's, while any small error on the measurement of the y 's introduces a large relative error on their difference. Similarly, expression (14), valid for a perfectly cylindrical head, leads to the same type of conclusions: when $\exp ik(L_1 - L_2)$ is equal to $+1$, which occurs if the two microphones are either at the same location or one wavelength apart, and when y is also equal to $+1$, again the numerator and the denominator of the fraction in the right hand side vanish, so that the expression becomes indeterminate. Consequently, if the two microphones are at a distance which is close to an integer multiple of the wavelength, the computation of Z requires to calculate the difference of almost equal numbers, and becomes inaccurate. Similarly, when $\exp ik(L_1 - L_2)$ is equal to -1 , that is when the distance between the microphones is a multiple of $\lambda/2$, the ratio y can only take the value -1 whatever measurement is done, and the same difficulties occur again.

The conclusion of this discussion is that, if the head is approximately cylindrical, the optimum distance between the microphones is roughly $\lambda/4$, which means in practice that different positions of the microphones are required for different frequencies. Of course, the optimum condition does not have to be matched exactly but, since in the experiment a large frequency domain is studied (from 50 to 2,500 Hz), more than two microphones are indeed needed; in this way, at each frequency one can choose a pair of microphones which approximately fulfills the optimum condition (for instance distance between $\lambda/10$ and $2\lambda/3$). In practice, we used different combinations, for example one in which the first microphone was not far from the reference plane, at 2.5 cm. from it, and two other microphones were respectively at 3 cm. and 10 cm from the first; one pair was used for

frequencies between 2,500 and 800 Hz, the other between 800 and 50 Hz. But we also used different versions, in particular one which was built in order to obtain more accuracy and to extend the frequency range, and which is described at the end of §4.1.

The choice of calibration cavities has similar requirements. The cavities are cylindrical tubes, with the same diameter as the cylindrical part of the head (15 mm.), except in the case discussed in §4.3; these reference tubes are closed at distances L' and L'' respectively. The theoretical value of the corresponding impedances can be found in ref. [25] for example. Now, the values of these lengths are to be chosen carefully; for example if L' is nearly equal to $\lambda/2$, that is if one of the reference cavities resonates and gets a very high acoustical impedance, the acoustical field inside the measurement head is almost the same as with the infinite impedance reference cavity; in other words y' is nearly equal to y_0 and no independent information is obtained from the two calibrations. Mathematically, in equations (6) and (7), one has to deal with the product of an infinite value of \bar{Z}' by a vanishing value of $(y' - y_0)$, which is indeterminate; in practice, the numerical value of the product is obtained with a very poor accuracy when one factor is very large and the other very small. The same problems obviously occur if L'' is chosen too close to $\lambda/2$, or an integer multiple of this quantity. Finally, if the difference $L' - L''$ is too close to an integer multiple of $\lambda/2$, the impedances \bar{Z}' and \bar{Z}'' will have similar values and the calibrations will not give independent information through the values of y' and y'' . Mathematically, this corresponds to situation where the system of equations (6), which is used to determine A and B , is degenerate (the determinant vanishes). Clearly, all these situations should be avoided.

Moreover, the large relative variations of impedances which take place when the frequency goes across an admittance resonance of the cavity (when the cavity length is close to $\lambda/4$) are also to be avoided. In theory, these large relative variations are exactly compensated in formula (8) by correlated large variations of the y 's, but again in practice this puts difficult requirements on the accuracy with which Z' and Z'' are computed theoretically, in particular in the precise evaluation of the losses. We attempted to avoid systematically these delicate situations, and found that a reasonable compromise is reached if L' is for instance of the order of $\lambda/3$, L'' of the order of $\lambda/6$. The consequence of this is that, depending on the frequency band studied, different sets of reference cavities are necessary. In our experiment, we used cavities respectively with lengths 100, 50, and 20, 10, 6 and 3 cm., and obtained a reasonable accuracy by choosing a suitable pair for each frequency band studied. The reference cavities were made of brass, as the measurement head itself, but the longest had thinner walls (1.5 mm.) for obvious practical reasons of price and weight.

The microphones we used first were high quality microphones (Bruel and Kjaer ref. 4134) but, after we gained more experience in the experiment, we found that cheaper microphones (Sennheiser KE4 100, fitted with preamplifiers) were perfectly suitable and gave identical results for impedance measurements. As emphasized at the end of § 2.1, this is because the only requirements on the microphones are reproducibility and linearity, while a frequency dependent gain or a difference of gain between the microphones does not introduce errors. They are coupled to the acoustical field inside the measurement head by small holes, 1 mm. in diameter, which in combination with the unavoidable cavity

in front of the microphones forms a system resembling a small Helmholtz resonator with high resonance frequency (more than 5 KHz). This clearly introduces frequency dependent amplitude and phase corrections, but we have seen in §2.1 that these are automatically corrected by our method.

The microcomputer (Tandon PCA 20) is fitted with a acquisition card (Analog Devices RTI 815) with a pair of A/D and D/A 12 bit converters suitably programmed to acquire two signals and generate one chirp simultaneously at a rate of 10 KHz, provided by an external clock. It can generate frequency chirps (duration 0.5 sec.) for the calibrations and the measurements themselves, store the time dependent signals over a time period of 1 second, Fourier transform these signals and eventually compute and store the value of y for each experiment. In addition, it computes the theoretical values of Z' and Z'' (closed cylindrical cavities) and, with the help of the stored values of the y 's inserted into (8), shows on its screen and eventually plots on a curve tracer the values of the modulus $|Z|$ of the impedance as well as its phase (see next section for examples of these plots).

The start of the acquisition is triggered by the microcomputer in conjunction with the beginning of the chirp. Because of the time duration necessary to the propagation of sound between the loudspeaker and the microphone, the signal is zero at the beginning; moreover, because the acquisition time is twice longer than the chirp duration, a good extinction of the signal at its end is obtained, even if high quality factor acoustical cavities are used. In this way the integrity of the signals is indeed stored, which allows us to compute the Fourier transform of the signals without any particular windowing. The Fourier transform code is a classical FFT program; the computations are performed on 8,192 points in floating point format on the two signals simultaneously. The delay induced by the multiplexer of the A/D converter is corrected by introducing a known phase shift on the second signal in order to obtain the Fourier transform of both signals with the same time origin.

In a real measurement, the successive operations which have to be made are the following:

(i) the calibration with the reference cavities in order to store the corresponding values of the y 's. Different chirps are used (frequency ranges 4000 to 800Hz, 1000 to 200 Hz and 250 to 10 Hz), depending on the length of the cavity. As mentioned before, a favourable situation occurs when the length of the cavity is between $\lambda/6$ and $\lambda/3$, so that long cavities are used with low frequency chirps, short cavities with higher frequencies. As a rule, all measurements are made at frequencies below the first resonance of the closed tube ($\lambda/2$) for reasons which have been discussed above; we give more detail on how this was done in practice at the end of §4.1. Step (i) is done once and for all for a series of measurements; it does not have to be remade until any important parameter, generally the temperature (see below) changes significantly.

(ii) the measurement itself is then made with the unknown impedance, most of the times with a series of three different chirps (the same as for the calibrations). When there is too much noise in the laboratory, an averaging procedure over 2, 4 or 16 successive chirps may be used. The whole measurement takes between 10 and 30 sec., depending whether averaging is made or not.

(iii) the computer then makes all calculations and plots the curves. Because the code has been carefully optimized, this step, without including the time of the calibrations, lasts

less than 3 minutes (it could of course be much shorter with a more powerful computer).

4. RESULTS

4.1 CALIBRATION FUNCTIONS

The first step was to measure and compute the three calibration functions A , B and y_0 , and to compare them with the predictions of the simple model developed in § 2.2. Figure 3-a shows the “theoretical” value of the calibration function A associated with a pair of microphones separated by a distance of 10 cm. ($L_1 = 2.5$ cm., $L_2 = 12.5$ cm.), that is the value obtained from relation (18-a); fig. 3-b shows the experimental value of this function obtained by using two reference cavities, with lengths 10 and 20 cm. respectively. Although the curves of figs. a and b are similar inside the “good” frequency domain where this pair of microphones is used for impedance measurements (from 200 to 800 Hz), there are also marked differences outside of this domain which are interesting to discuss, even if they do not affect the value obtained for Z . The first perturbation occurs around 850 Hz, which corresponds to a $\lambda/2$ resonance of the 20 cm. calibration cavity (see discussion of §3). It is clear that this perturbation is entirely artificial, in other words that it can not occur in the real values of A but actually comes from an incorrect theoretical evaluation of the impedance the reference cavity: because the function A characterizes only the physical properties of the head with the two microphones, there is no reason why it should exhibit any sharp variation at a frequency which depends on the length of the reference cavity. It would therefore be a good approximation to make a smooth mathematical interpolation between the two sides of this resonance, in other words to merely wipe it out. In the case of the perturbations around 1700 Hz, the situation is more complicated because, in addition to resonances of the reference cavities, one falls at a “forbidden” frequency where the distance between the microphones is $\lambda/2$. This is probably why the experimental curve becomes so irregular, and it is less clear how it could be simply smoothed. Nevertheless we emphasize again that this is by no means necessary: if the sharp variations of the calibration function are shown for full information, they remain irrelevant for our measurements because, for each frequency domain, we use a pair of microphones and of calibration cavities for which the measured values of A have only smooth variations; consequently, the sharp irregularities do not affect the experimental results at all.

Indeed, the effect of changing the pair of measurement microphones is illustrated in figures 4-a and b, which show similar curves for a pair of microphones which are at a smaller distance (3 cm.; $L_1 = 2.5$ cm., $L_2 = 5.5$ cm.); figure a gives the theoretical predictions (equation (18-a)), figure b the experimental results. One can see that, now, experiment and theory match much better at high frequencies, as expected, and in fact the “forbidden” frequency is too high to be seen in the plot. On the other hand, the results obtained with this pair of microphones would be less precise at low frequencies (because the values of the ratios y 's would be too close; see discussion of preceding section).

Figures 5-a, b show similar results for the calibration function B and the same pair of microphones as that of figure 4 (microphones 3 cm. apart; for the pair of microphones which are 10 cm. apart, similar problems to those of fig. 4 would occur) ; figures 6-a and 6-b the corresponding results for y_0 , in addition, for more clarity, fig. 6-c shows the ration between the experimental and theoretical values. On the latter figure, one notices a

large discrepancy associated with to a peak around 1550 Hz, which is the frequency where the value of y_0 goes through a minimum ($k_r L_2 = \lambda/4$); this is due to the fact that the theoretical value has a more pronounced minimum than in its experimental counterpart; it indicates that, at frequencies where the losses become dominant, it is difficult to account precisely for their values so that non-negligible errors are introduced. Meanwhile, as we see below, it is important to have a good control of the values of this calibration function: small variations of this function can result in relatively large variations of the result for the impedance measurement, especially of the height of the peaks which obviously depends critically on the value of y_0 . We therefore have here a case where using, as we do, the experimental values of the calibration functions can significantly affect the measured value of the impedance (at least for frequencies falling inside the peak of fig. 6-c).

The values measured for the calibration function are changed if the temperature, that is the sound velocity, changes, in particular because the frequency at which y_0 goes through a minimum is modified. For example, when the temperature goes from 25 to 27 degree Celsius, we could clearly see on our plots modifications of the values of the calibration function, although they would be harder to see on small figures so that we do not show them here. The measured values also depend on the perturbations of the acoustical field by the microphones, as one can check by comparing figures 6-b and 7, which have been obtained in the same experimental conditions, except that the Bruel and Kjaer microphones were replaced by a pair of Sennheiser microphones for the latter figure. This clearly indicates that the influence of the microphones, which have non-zero admittances, plays a non-negligible role in the experiment and should not be ignored for accurate measurements.

After studying the values of the calibration functions in various circumstances, we ended up using two different combinations of microphone positions and calibration cavity lengths which, we feel, correspond to two reasonable compromises:

set-up I:

Frequency range	L_1	L_2	Lengths of the reference cavities
50 – 200 Hz	2.5 cm	12.5 cm	100 and 50 cm
200 – 800 Hz	2.5 cm	12.5 cm	20 and 10 cm
800 – 2500 Hz	2.5 cm	5.5 cm	6 and 3 cm

set-up II, shown in figure 2, which requires 4 microphones but gives a larger frequency range (at low frequencies because there is a pair of microphones at a larger distance, at high frequencies because the first microphone is closer to the reference plane:

Frequency range	L_1	L_2	Lengths of the reference cavities
10 – 200 Hz	1 cm	31 cm	100 and 50 cm
200 – 800 Hz	1 cm	11 cm	20 and 10 cm
800 – 4000 Hz	1 cm	4 cm	6 and 3 cm

The flexibility which one has to choose the pair of measurement microphones and the set of calibration cavities is an important feature of the method.

4.2 IMPEDANCE MEASUREMENTS

We now discuss the results of real impedance measurements, made with the help of the calibrations functions obtained experimentally by the method of the preceding section.

4.2.1 Cylindrical cavities

As a test of the accuracy of the method, we first decided to measure the impedance of closed cylindrical tubes, which are actually very similar to the cavities used for the calibrations. A good reproduction of the theoretical values of the heights of the impedance peaks, of its minimums, as well as of the phases, gives indeed a severe check of the validity of the method and of its ability to give a sufficient dynamic range. Also, what can be done is to use the data of the same measurements, but formula (14) instead of (8) for the calibration functions. If less plausible results are obtained, it is clear that the calibration method plays a significant positive role.

Figure 8 shows the results obtained precisely in this way, that is by using theoretical values for A , B , and y_0 , for a closed tube 100 cm. in length; then, our method becomes merely equivalent to that of references [12] to [14] for example. These results should be compared to those of fig. 9-a where the experimental values of the calibration function were used; as a reference for the eye, a theoretical curve showing the height of the peaks has been drawn in broken line (in both cases, the set-up I was used). The better regularity of the peaks of fig. 8 shows the clear improvement arising from the calibration procedure. Fig 9-b shows the same data, but in logarithmic scale: one notices that the impedance peaks and notches are obtained with a comparable accuracy, which is not necessarily the case if one uses the capillary method of impedance measurements. The accuracy for the determination of impedance peaks is of the order of a fraction of a Hertz (since the duration of the signal is one second, the calculation of FFT gives independent points separated by one Hertz, but we interpolate numerically between these points in order to improve the accuracy; in particular, the points at which the phase goes through zero can be measured with an accuracy of 0.1 Hz or even less). We then checked that the same results are obtained when the intensity of the chirps generated by the loudspeaker is multiplied or divided by two; this ensures that non-linear phenomena, such as turbulence, do not play an important role in our experimental conditions.

4.2.2 Real musical instruments (clarinets)

For doing measurements with real instruments, we moved the measurement head so that the musical instrument could be held vertically, which allows a clarinettist to make the fingerings more naturally; this is useful even if each measurement lasts only a few seconds. A cylindrical tube of 65 mm. length and 15 mm. internal diameter, which corresponds to the volume of a usual mouthpiece (12.5 cubic centimeters) was inserted between the measurement head and the barrel. In this way, one measures the impedance at the tip of the reed to a good approximation (at least when the impedance is high, that is in the vicinity of impedance peaks, but this is what is mostly relevant to the motion of the reed and therefore for the functioning of the instrument). Figure 10 shows the impedance curve of a low cost clarinet (Noblet) with a low E fingering (lowest note of the

instrument). Although obtained in a much shorter time, this curve compares well with curver obtained with impedance bridge based on the use of capillary tubes. We have used our impedance bridge to make a large number of impedance measurements on various fingerings for clarinets of different origins, as well as bass clarinets. This brings us to the subject of measuring the impedance of resonators with diameters which are different from that of the cylindrical head.

4.3 DIFFERENT DIAMETERS

In a second step, and in order to have a completely independent check of the validity of the calibration method, we decided to use the same measurement head to determine impedances of resonators with a different diameter. While the use of the theoretical formula (17), or any similar formula obtained from the mathematical study of the propagation of acoustical waves in cylindrical ducts, does not allow to change the diameter of the measured impedance, our method does not have this limitation: the calibration procedure automatically takes care of all effects related to the propagation of sound inside conical sections or even in principle diameter discontinuities (provided the basic assumption of §2 remains valid: one must not have more than one single transverse mode in the region of the reference plane). Of course, this requires the use of other values for the calibration functions A , B and y_0 , which have to be obtained initially by using other reference cavities that have a diameter equal to the new diameter chosen for the neck of the unknown impedances.

In practice, it is useful to avoid discontinuities which would reflect strongly the acoustical wave in a way that does not depend much on the calibration (or measured) impedances. This would clearly limit the sensitivity by giving close values to all y ratios. Moreover, sharp discontinuities would create intense local, non propagating, modes which are unwanted close to the reference plane. In other words, one should try to ensure good adiabatic conditions for the propagation of the waves everywhere inside the measurement head. Consequently, we inserted a conical section between the unmodified cylindrical part of the measurement head and all impedances (including the infinite impedance of calibration); this conical section is 2 cm. long and its diameter varies from 15 to 6 mm. In the reasoning of §2.1, it should be considered as the final part of the measurement head.

Figure 11 shows the new values of the calibration function y_0 , which differ substantially from the values shown in figure 6-b, although the distance between the microphones is the same (namely 3 cm.); this is due to the effect of the added conical section. Figure 12-a shows the results obtained by measuring the impedance of a cylindrical tube (length 100 cm., diameter 6 mm.). As a point of comparison, fig. 12-b shows the results obtained with the same data, but by using the calibration function of §4.2, which puts into light the importance of adequate calibration functions. Figure 13 shows the impedance peaks of a Leblanc bass clarinet. An interesting feature seems to be the occurrence of a second allowed band, beyond the forbidden band which lyes between the usual cutoff frequency (around 1100 Hz) and another limit frequency (around 1800 Hz).

4.4 REFLECTION FUNCTIONS

From the same data, one can also extract the reflection function $R(t)$, which is given

by the inverse Fourier transform of the function [26]:

$$\tilde{R}(\omega) = \frac{\bar{Z}(\omega) - 1}{\bar{Z}(\omega) + 1} \quad (19)$$

where $\bar{Z}(\omega)$ is the dimensionless impedance (the ordinary impedance divided by ρc). Mathematically, the information contained in both functions is the same, but physically the reflection function is especially useful in models of the functioning of woodwind instruments in the time domain [26] [27] [28].

With our experimental setup, it is easy to add at the end of the code of the microcomputer the few lines which are necessary to compute $\tilde{R}(\omega)$ and then to Fourier transform to obtain $R(t)$; the whole calculation takes about one extra minute. To avoid parasitic oscillations introduced by a brutal frequency domain truncation, we multiply $\tilde{R}(\omega)$ by an apodisation function which is equal to one for low frequencies up to 2200 Hz and then decreases progressively to zero until 2500 Hz. Figure 14 shows the reflection function of the cylindrical tube used in figure 9. It compares favourably with reflection functions computed theoretically [28] although some parasitic oscillations, due to the limitation of our frequency domain to 2200 Hz, are visible.

5. CONCLUSION

The basic principle of the method that we have used for acoustical impedance measurements is not new; it relies on ideas which go back to the last century, as we have recalled in the introduction. The method is actually still used for measuring acoustical impedances of various absorbing materials which can be put inside a cylinder with a diameter of a few centimeters (see for example refs. [10] to [13]). In the present article, we have addressed a different variant problem, since we are interested in the impedance of resonant cavities. This raises more severe technical requirements because of the larger dynamic range which is required to reproduce impedance peaks and dips correctly.

We have shown that it is indeed possible to meet these requirements by the use of a calibration procedure which allows one to measure the coefficients A , B and y_0 , instead of computing them from the knowledge of the geometrical characteristics of the acoustical system (presumably within the frame of some simplified model to take into account the perturbations of the acoustical field by the microphones). Besides geometrical imperfections of the measurement head or of the microphones themselves, the method eliminates several sources of systematics, as discussed above (finite admittance of the microphones, effect of the small cavities in front of the microphones, etc.). Moreover, the method includes a large flexibility, not only on the choice of the distance between microphones, but also on the selection of calibration impedances. Indeed, by using different sets of calibration cavities for different frequency ranges, we have systematically avoided resonances, where one has to calculate theoretically the acoustical losses either in these cavities or in the measurement head; this precaution is useful because these losses are in practice not easy to evaluate with precision. All this calibration procedure is done once and for all, and stored initially in the computer; it does not have to be repeated for each measurement. Moreover, the possibility of checking from time to time the values of the calibration functions, which

depend critically of many parameters of the experiment (the exact position of microphones, their nature, the temperature, etc.) can be seen as another useful feature of the method: one keeps better control in this way of perturbations which of course also occur in other methods but remain unknown. On the other hand, a limitation of our method is that it requires that the structure of the acoustical wave be plane in the vicinity of the reference plane (more precisely, one assumes a single transverse mode); the excitation of local modes in this plane would perturb the basic principle on which the measurement relies, and actually would raise questions about how to define one single acoustical impedance in this plane. Nevertheless, this restriction applies only to the reference plane: as we have seen, the excitation of local non-propagating modes at other locations of the measurement head (near the microphones for example) is automatically compensated.

When these precautions are taken, we have seen that satisfactory results are obtained in the case of impedance measurements for closed cylindrical cavities, which have high and narrow resonance peaks, and in particular that the heights of the peaks are close to their correct values. The comparison between figures 8 and 9-a shows that more irregular peak heights would be obtained without the calibration procedure, which illustrates its usefulness (if more than the mere position of the peaks is expected from the measurements). Another convenient feature is the possibility of studying acoustical systems of various diameters, provided the calibrations are made with similar cylindrical cavities.

In terms of time taken by the measurement and sensitivity to the ambient noise, the method seems to be compatible with the environment that can be found in the vicinity of the workshop of instrument makers. The performances could be improved by using faster computers, such as the ones that have become available recently on the market for a reasonable price and work at a clock frequency of 33 MHz with 32 bit processors. What remains to be seen, however, is whether or not it is possible to find in impedance measurements an useful guide for either controlling the quality or improving clarinets for example. Only experience will decide this, but we nevertheless feel that, because there is clearly a lot of information in the impedance curves or in the reflection function (including in the phases which are often overlooked), there is a good chance that these measurements will prove to be a useful tool in the future.

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FIGURE CAPTIONS

Fig. 1: Scheme of the setup considered in the general discussion of §2.1: the measurement head, not necessarily cylindrical, is fitted with two microphones M_1 and M_2 , which may have non zero admittances and be coupled to the acoustical field in different ways (e.g. through holes having different diameters or lengths); they may also have different characteristics. What is important for the impedance measurements is that the acoustical wave obeys purely outgoing boundary conditions at the output of the unknown impedance Z (this is symbolized by the arrow on the right of the figure). The reference plane where this impedance is defined is shown in dotted lines. It assumed in the discussion that, in the vicinity of this plane, the acoustical wave contains only one transverse mode.

Fig. 2: Sketch of the experimental setup; it consists in a loudspeaker, four microphones, a microcomputer fitted with D/A and A/D converters, and three reference impedances (one of them infinite), not shown, which are connected in succession in place of the unknown impedance Z . As in figure 1, the reference plane (where the impedance is measured) is shown in dotted lines. This figure corresponds to the set-up called II at the end of section 4.1: it includes four microphones situated at distances 1, 4, 11, and 31 cm. from the reference plane respectively.

Fig. 3 shows a calibration function A for two microphones 10 cm. apart. The upper curve shows the modulus of the function, the lowest part the phase. Fig. 3-a shows the "theoretical" value of A given by formula (18-a) and fig. 3-b the experimental value obtained by using three reference cavities (formula (7)). In the frequency domain where these microphones are used (from 200 Hz to 800 Hz), one notices an acceptable, but not perfect, agreement between the two values; the differences show the usefulness of the calibration procedure. At 1700 Hz, a "forbidden" frequency occurs because the microphones are separated by a half-wavelength so that they do not give really independent information; this explains the complete disagreement from the "theoretical" value, but this perturbation is of course of no relevance to our experiment. See text for a discussion of the fast variations around 850 Hz.

Fig. 4: calibration function A (amplitude and phase) for a pair of microphones 3 cm. apart (as in fig. 3, the theoretical and experimental values appear in figs. a and b; the amplitude and the phase are plotted in the same way). Except that the distance between the microphones has been changed, all parameters are the same as in figure 3. This pair of microphones is used in the band 800 Hz to 2500 Hz. Similar remarks to those of the caption of figure 3 can be made, but the "forbidden" frequency now occurs at too high a frequency (5700 Hz) to be seen on this diagram.

Fig. 5: Results obtained for the calibration function B , plotted in the same way as in figures 3 and 4.

Fig. 6: Results obtained for the calibration function y_0 ; figs a and b are similar to figures 4, 5 and 6, while figure c is gives the ratio between the experimental and theoretical values. One notices a peak around 1550 Hz, corresponding to a frequency at which M_2 is at a distance of $\lambda/4$ from the reference plane. This calibration function is essential in the

determination of the heights of the impedance peaks, and the effect of the peak is indeed included in this determination; for more details, see text.

Fig 7: calibration function y_0 obtained in conditions similar to those of figure 6, except that the microphones are now Senheiser microphones; the curve looks more similar to that of fig. 6-b, which means that the perturbations of the acoustical field by the microphones (which are smaller) is reduced.

Fig. 8: values obtained, in linear scale, for the impedance of a cylindrical closed tube of length 100 cm, by using the theoretical values of the calibration functions instead of experimental calibrations; the upper curve gives the modulus of the impedance, the lowest the phase. The results are significantly less satisfactory than those of fig. 9, where the whole calibration procedure of our method has been used; note in particular that the resonance frequencies, and not only the heights of the peaks, are different; the phases are also strongly distorted at high frequencies.

Fig. 9: figure (a) shows, in linear scale, the results of an impedance measurement performed with the same closed cylindrical tube of length 100 cm than for fig. 8, but here with three auxiliary calibrations. For comparison, the theoretical heights of the peaks are shown with a broken line; this provides a severe test of the accuracy of the method, in particular of the quality of the calibration function y_0 . Figure (b) shows the same results for the modulus of the impedance on a logarithmic scale, which is more appropriate when one is interested in admittance peak determination. Because the reference cavities are not resonant in the frequency domain where they are used, these admittance peaks are determined with an accuracy comparable to the impedance peaks.

Fig. 10: Impedance curve of a Noblet clarinet, with a low E fingering. The data, modulus and phase, is obtained in a total measurement time of 10 seconds. An averaging over 8 chirps was taken in each of the two frequency bands in order to improve the accuracy, but the results are actually almost as good without averaging.

Fig. 11: New calibration function y_0 obtained for reference cavities of a different diameter (6 mm.) and with the insertion of a 2 cm. conical section at the end of the measurement head. Comparing this figure to fig. 6-b illustrates the acoustical effects of the conical section. The distance between microphones is 3 cm.

Fig. 12-a: measured impedance curve of a 100 cm. cylindrical cavity of internal diameter 6 mm; the theoretical curve giving the heights of the peaks is shown in broken line for comparison. Fig. (b) shows the values of the impedance that are obtained from the same data, but with inadequate calibration functions (those of fig. 9 and 10, obtained for a different diameter).

Fig. 13: Impedance curve of a Leblanc bass clarinet. One notices, beyond the first "forbidden band" between about 1100 and 1800 Hz., the appearance of a second "allowed band".

Fig. 14: Reflection function of a cylindrical closed tube. In the absence of dissipation, this curve would be a delta function.

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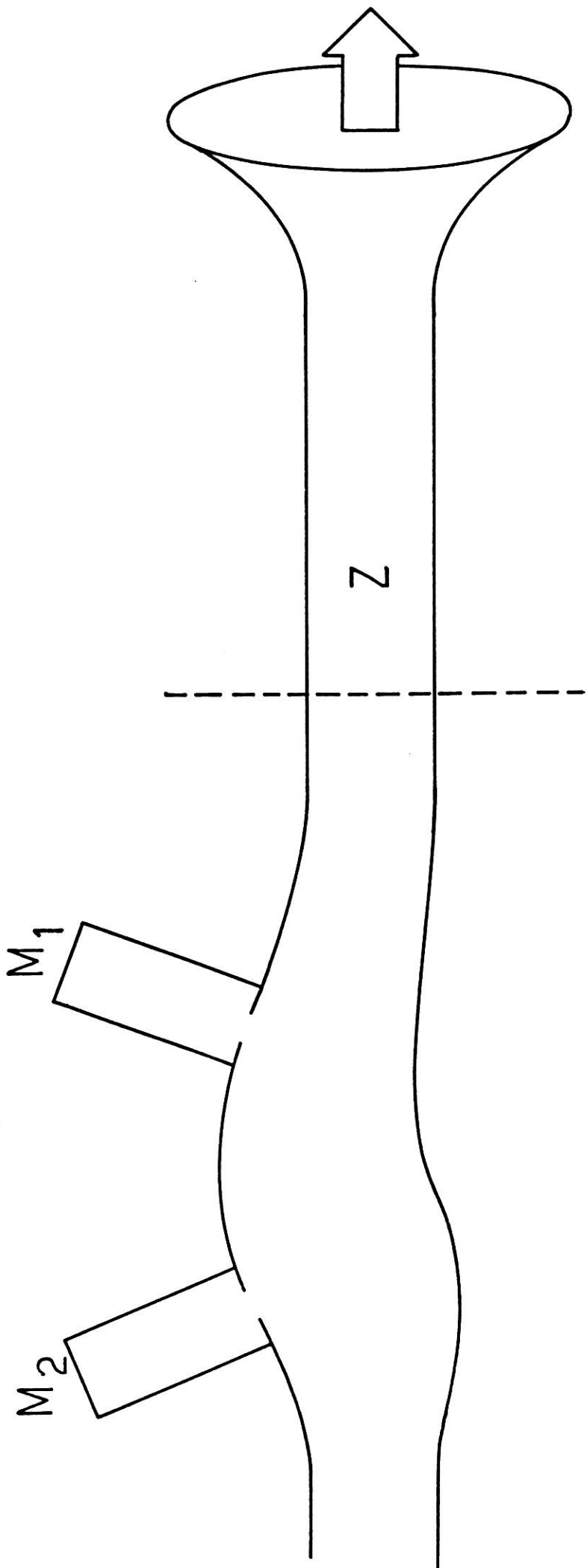


Fig. 1

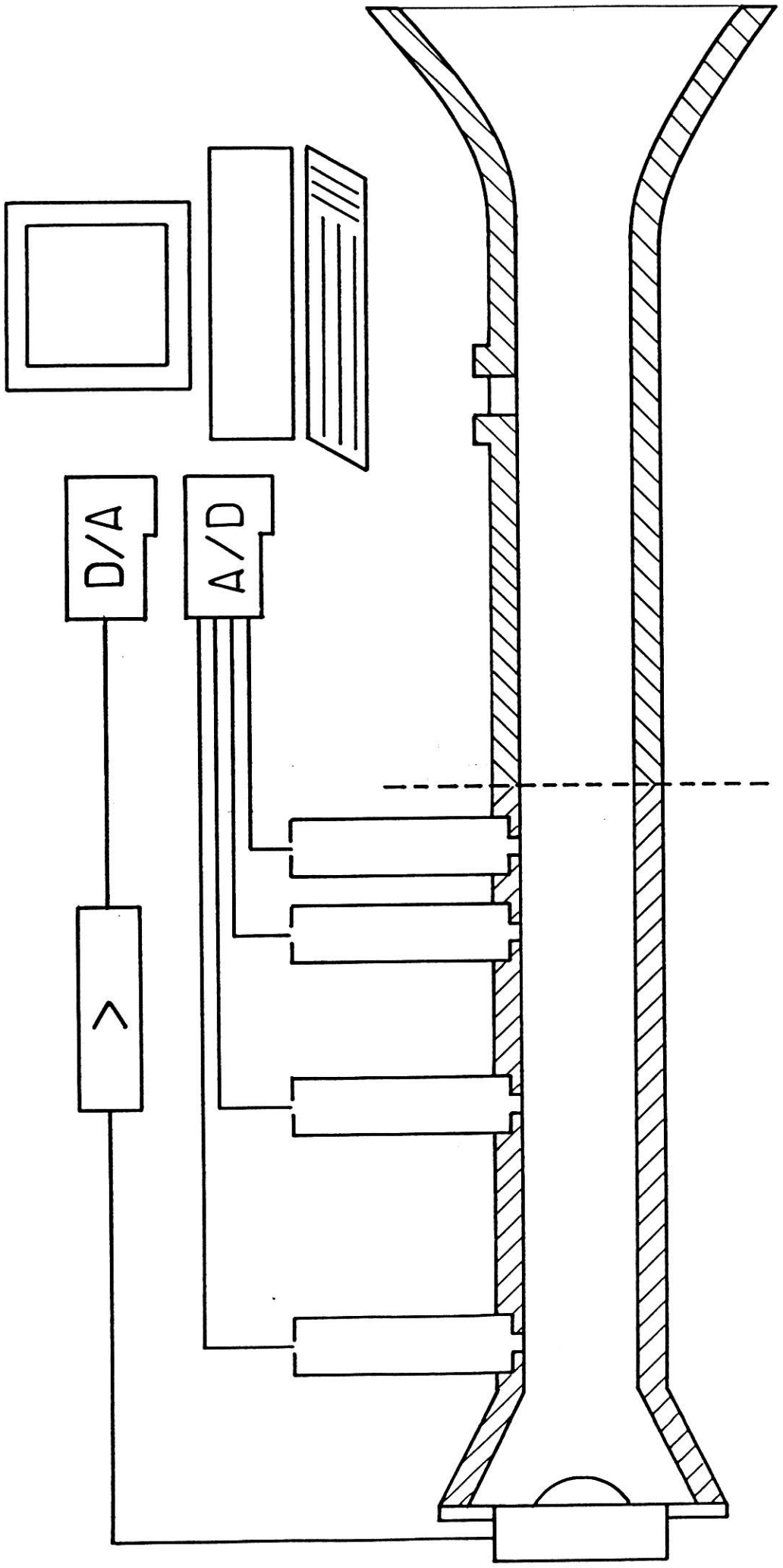


Fig 2

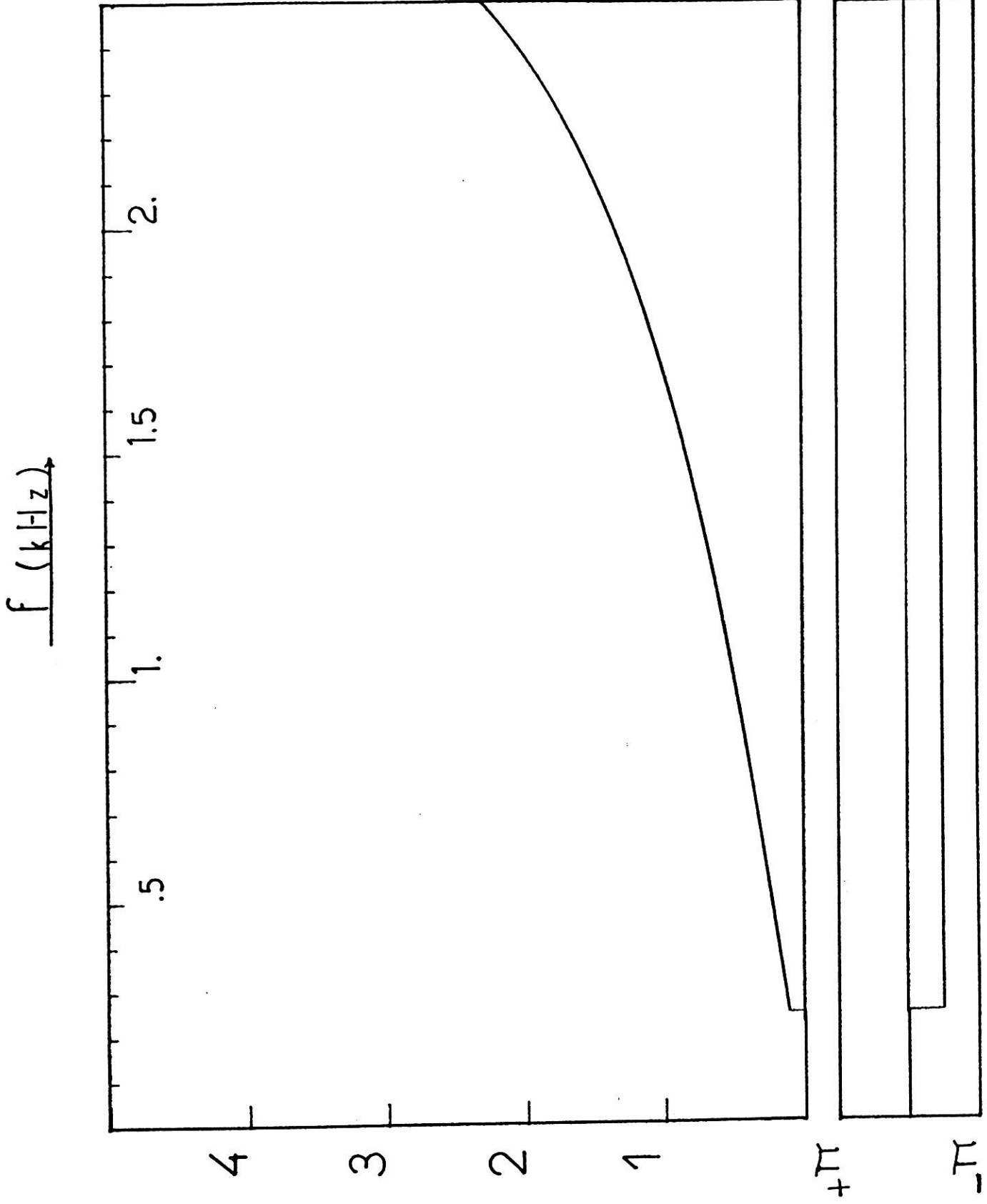


Fig 3a

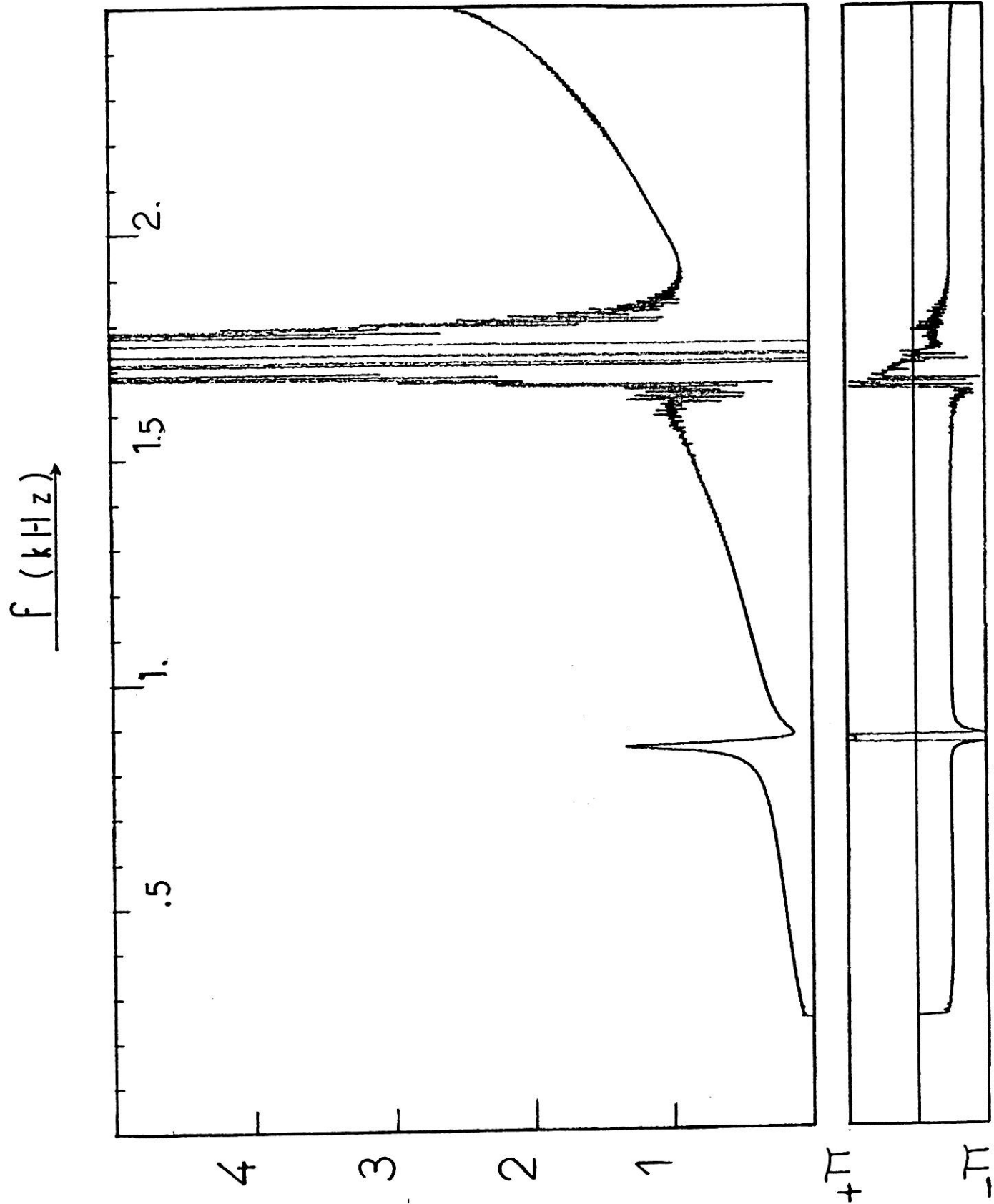


Fig 3-b-

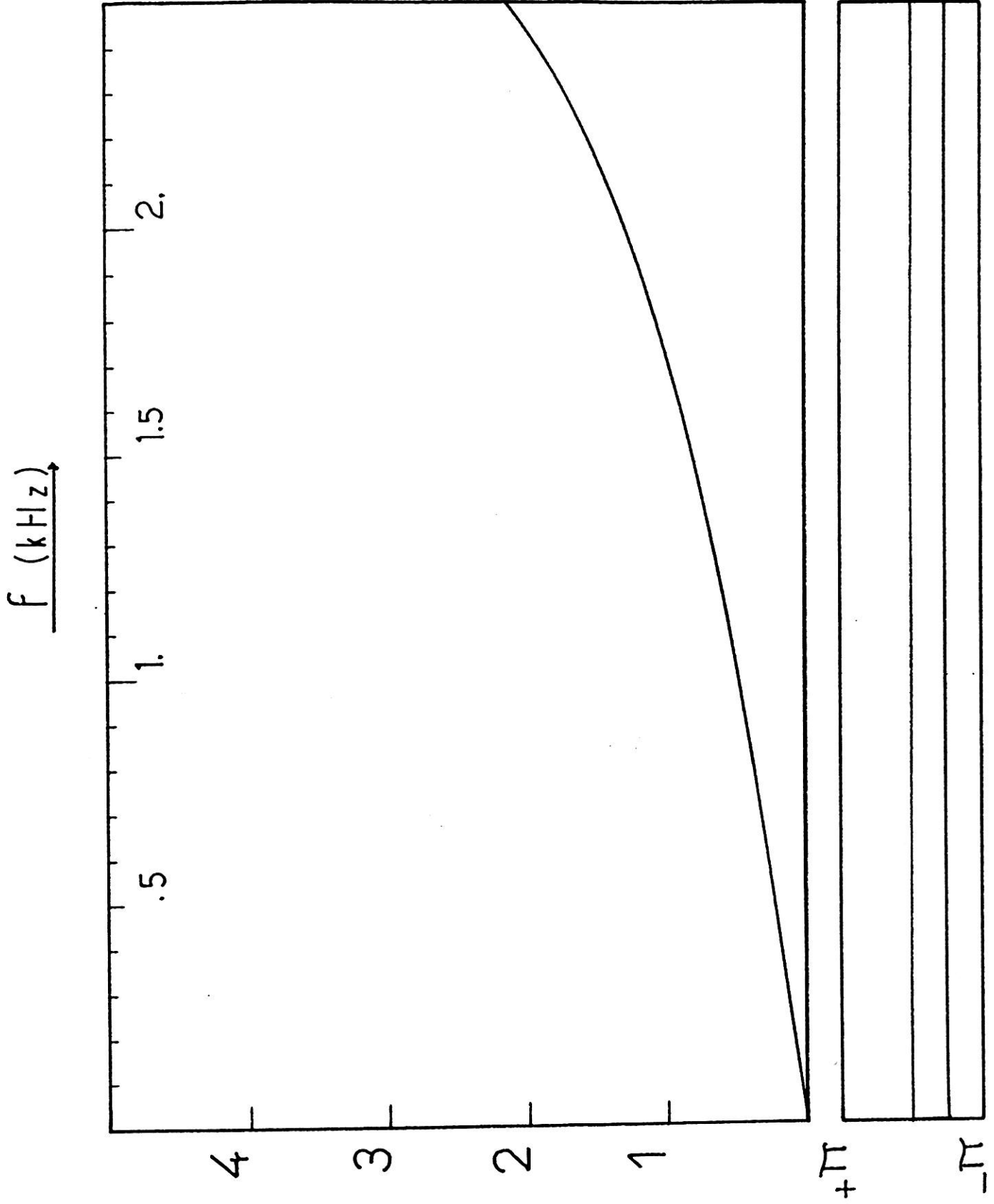


Fig 40

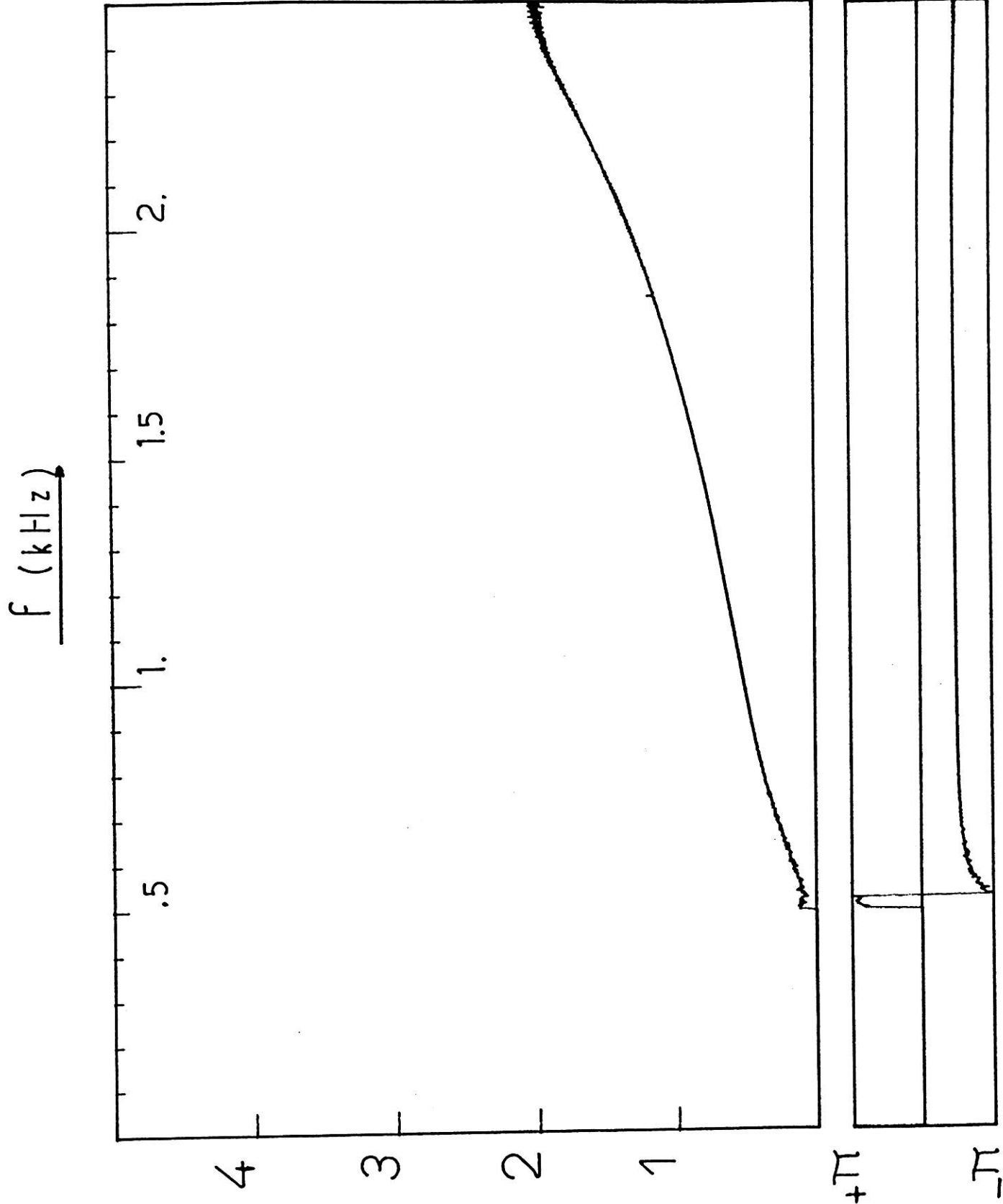
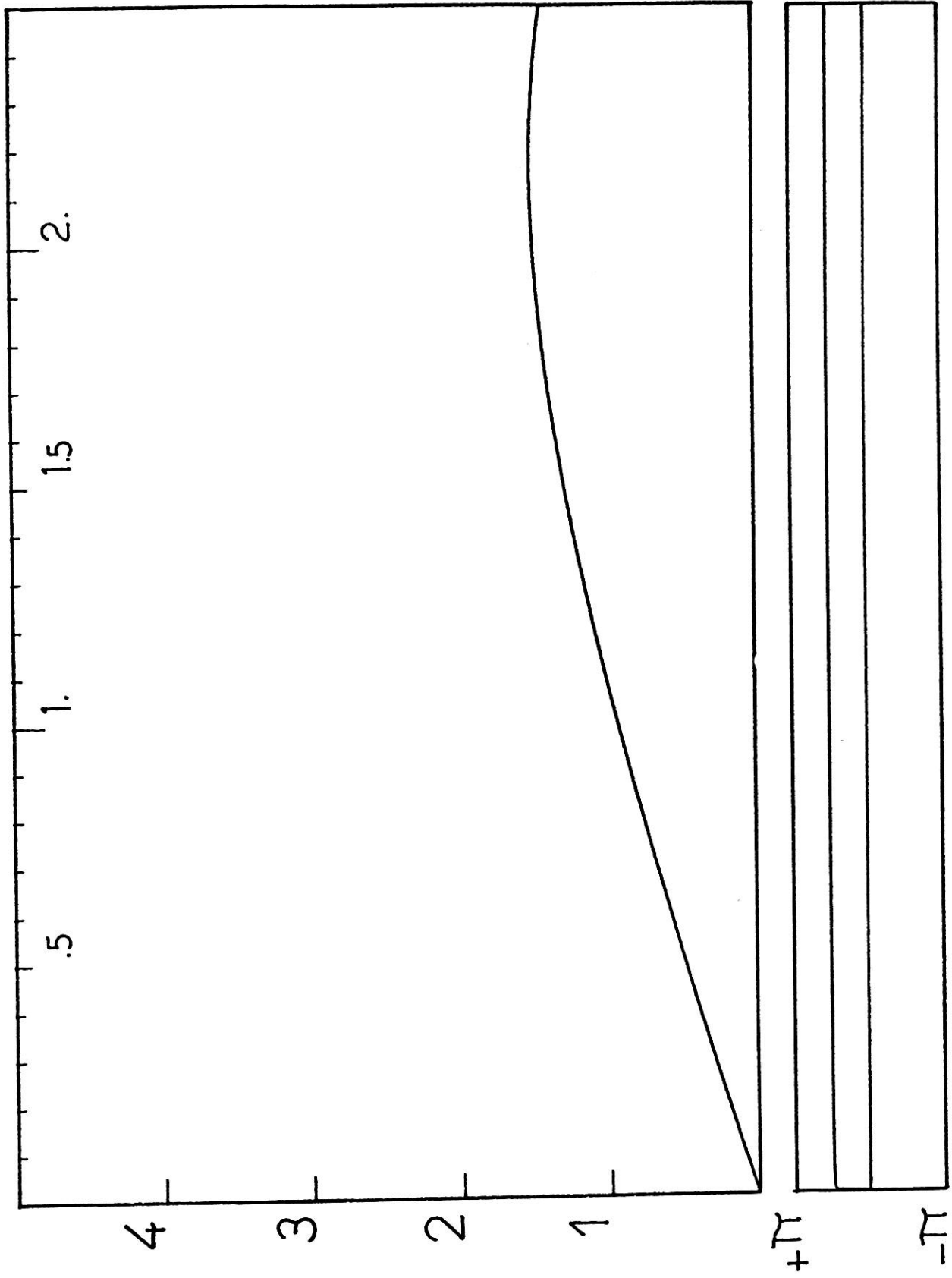


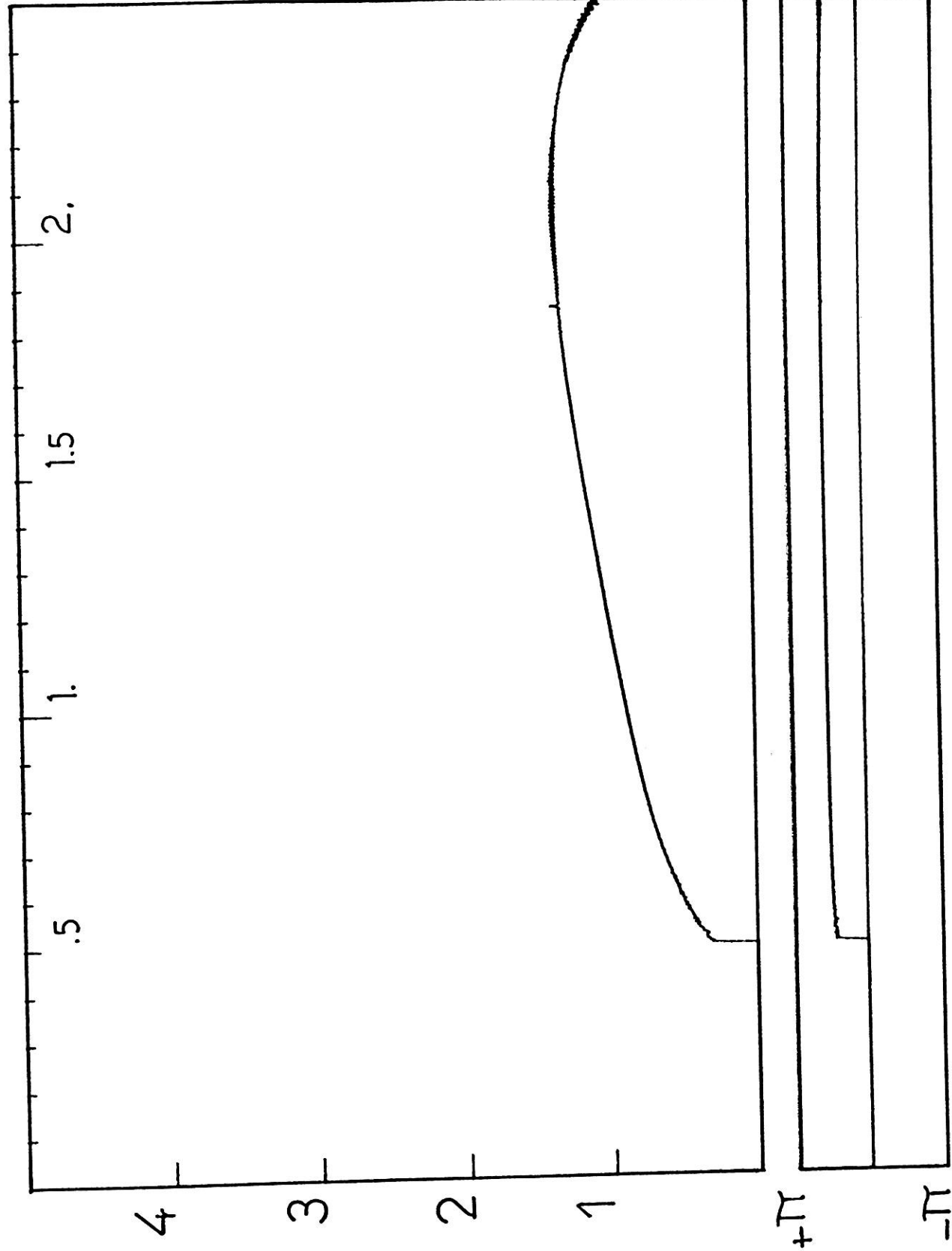
Fig 4-b

f (kHz)



705a

f (kHz)



h) 5b

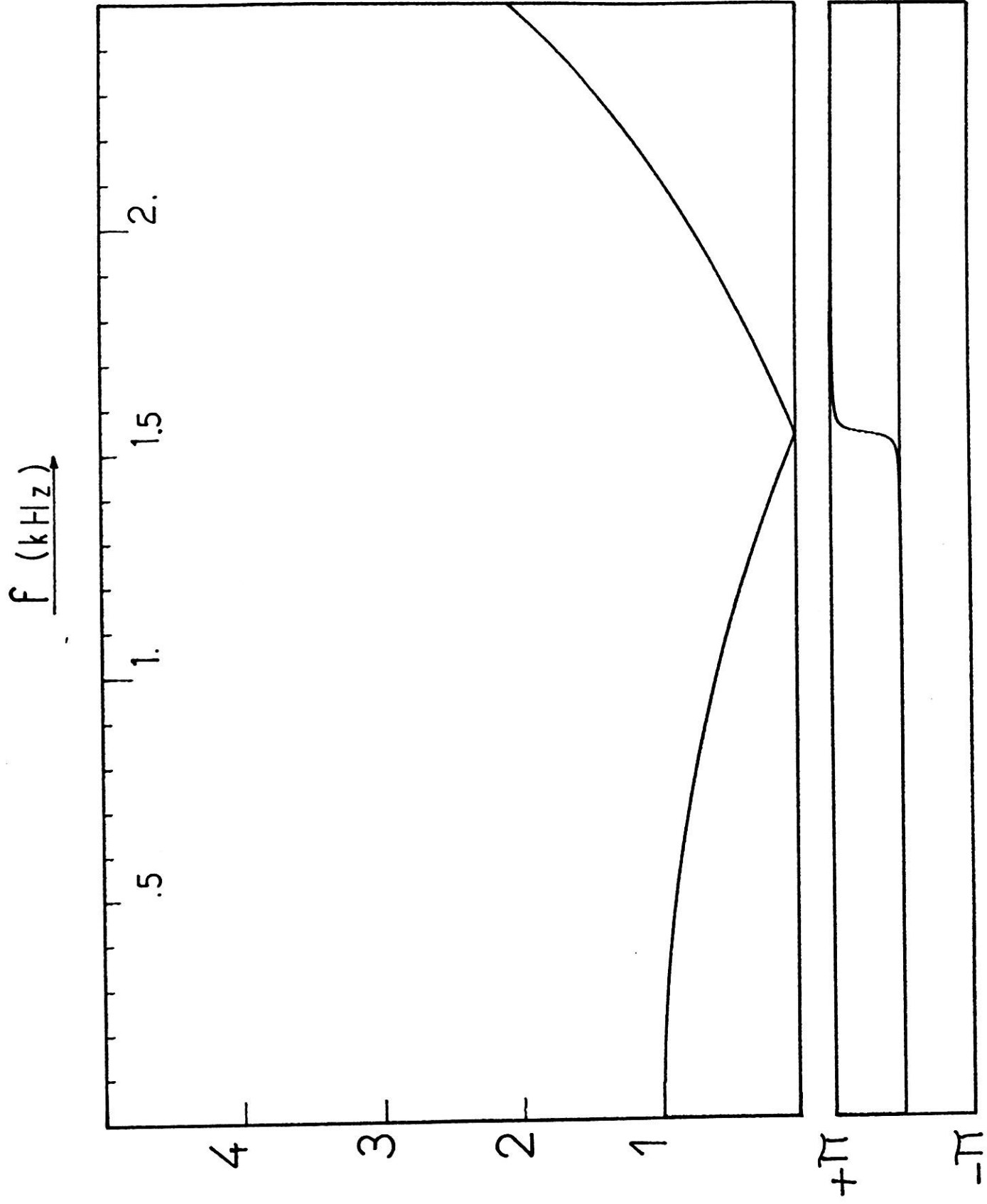
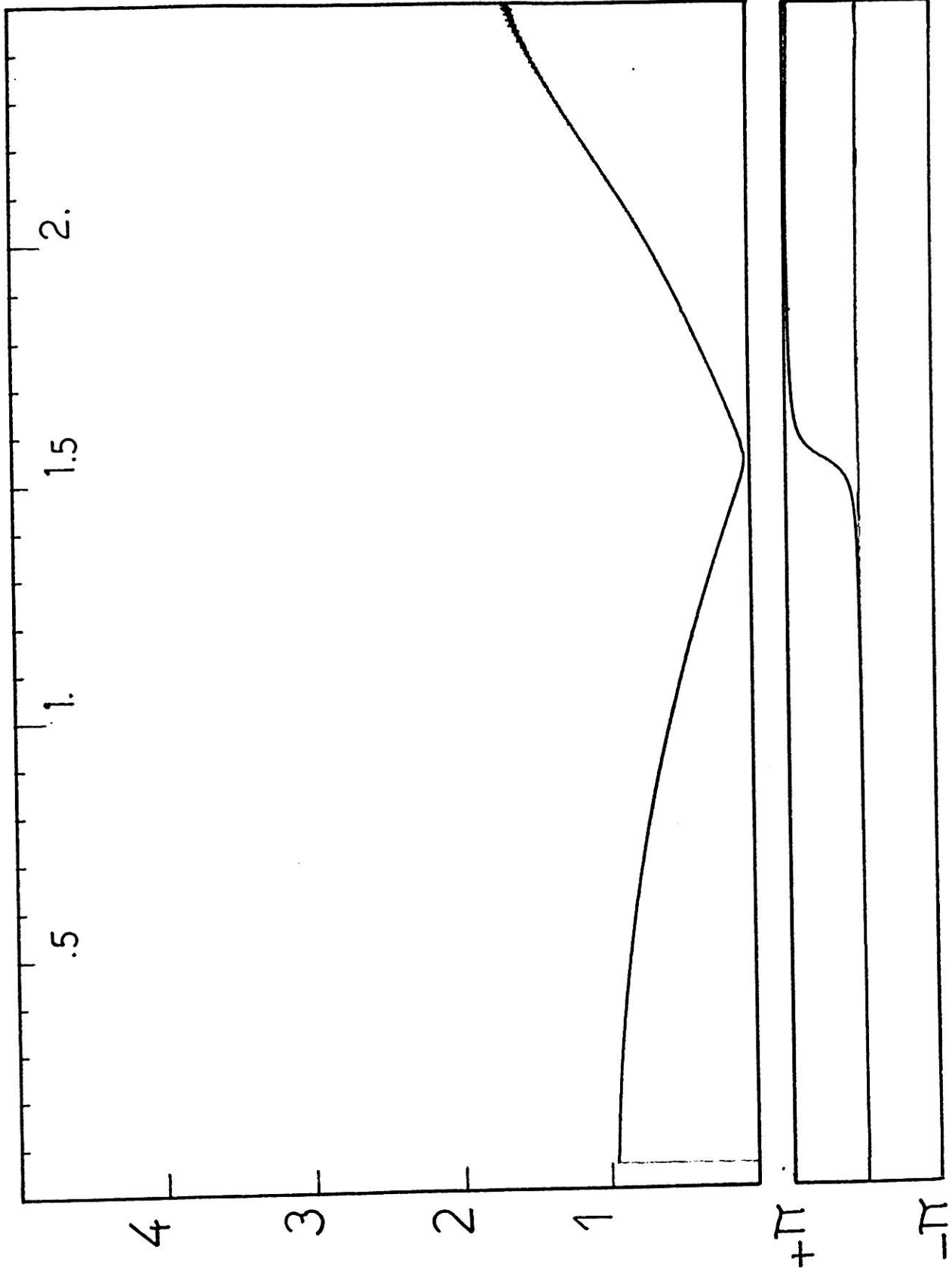


Fig 6a

f (kHz) \rightarrow



f (kHz)

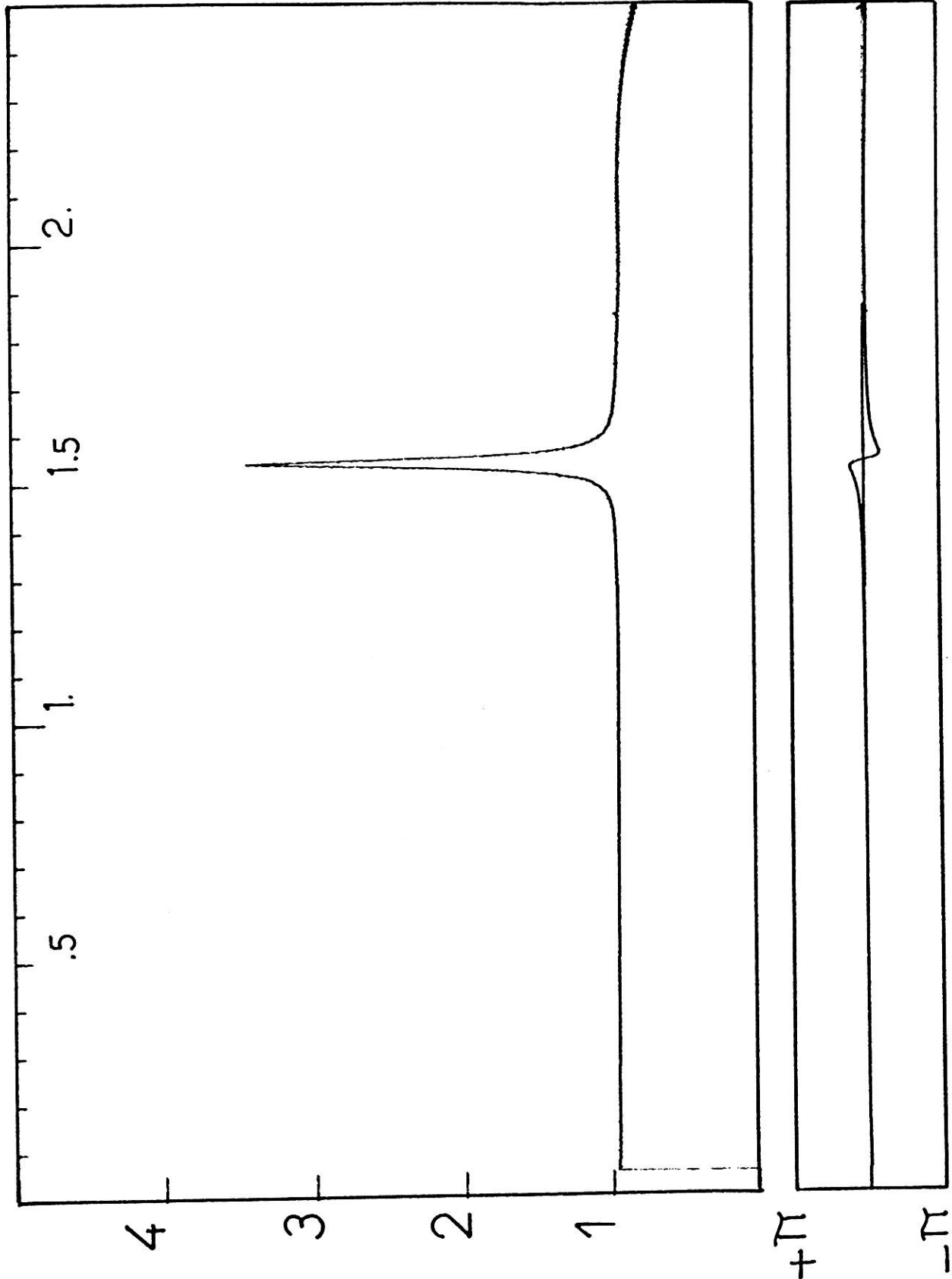


fig 6c

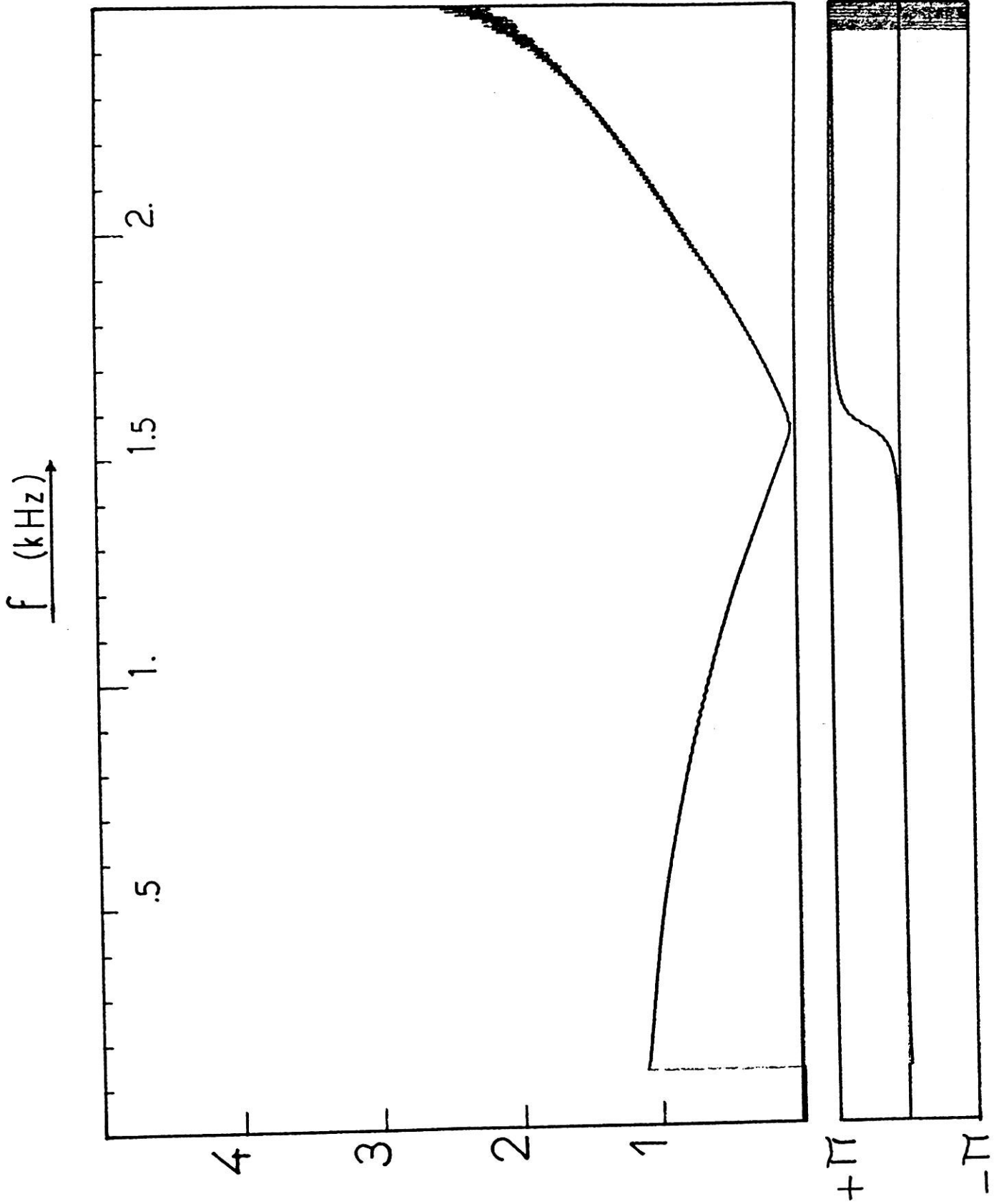
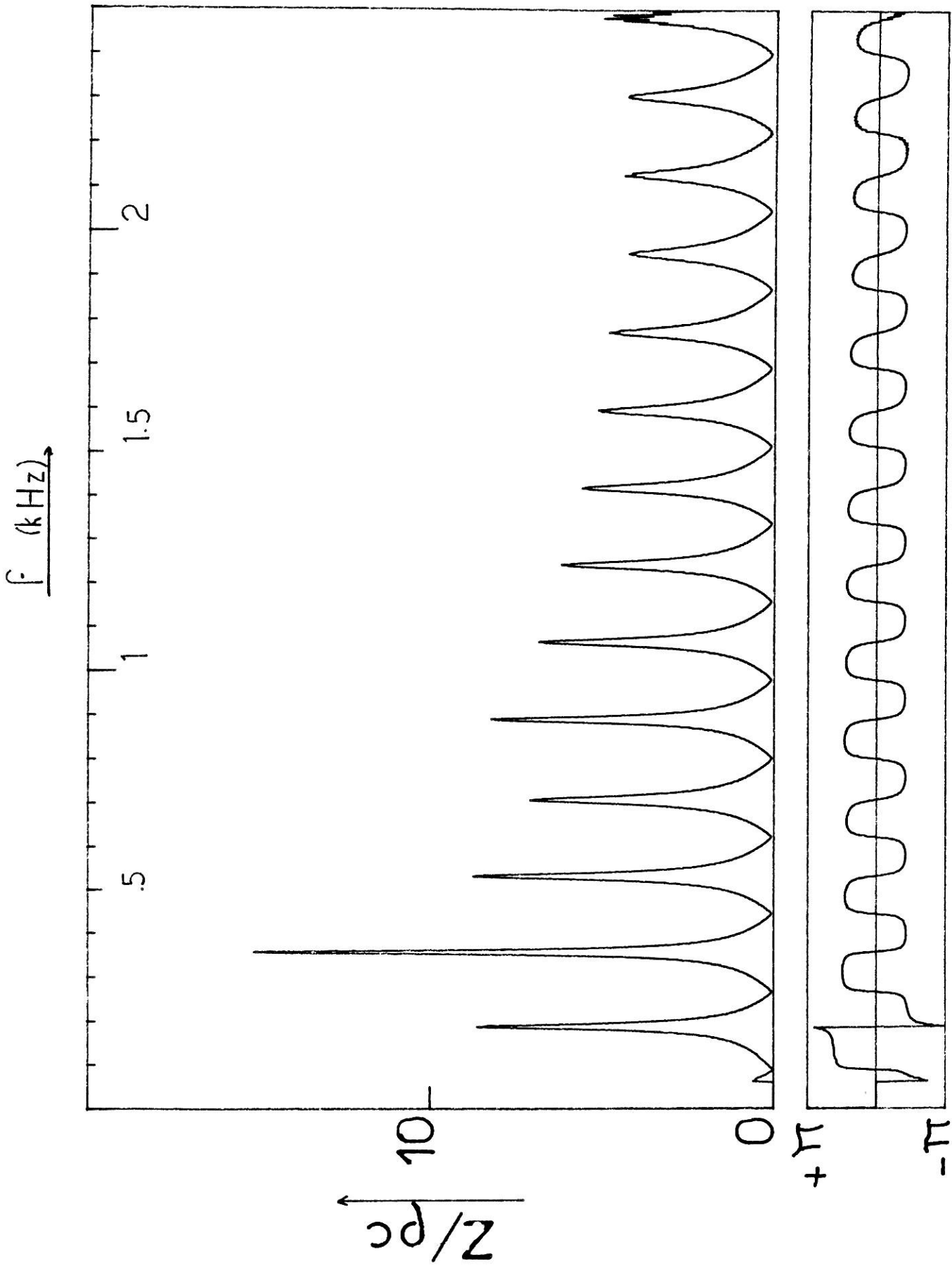
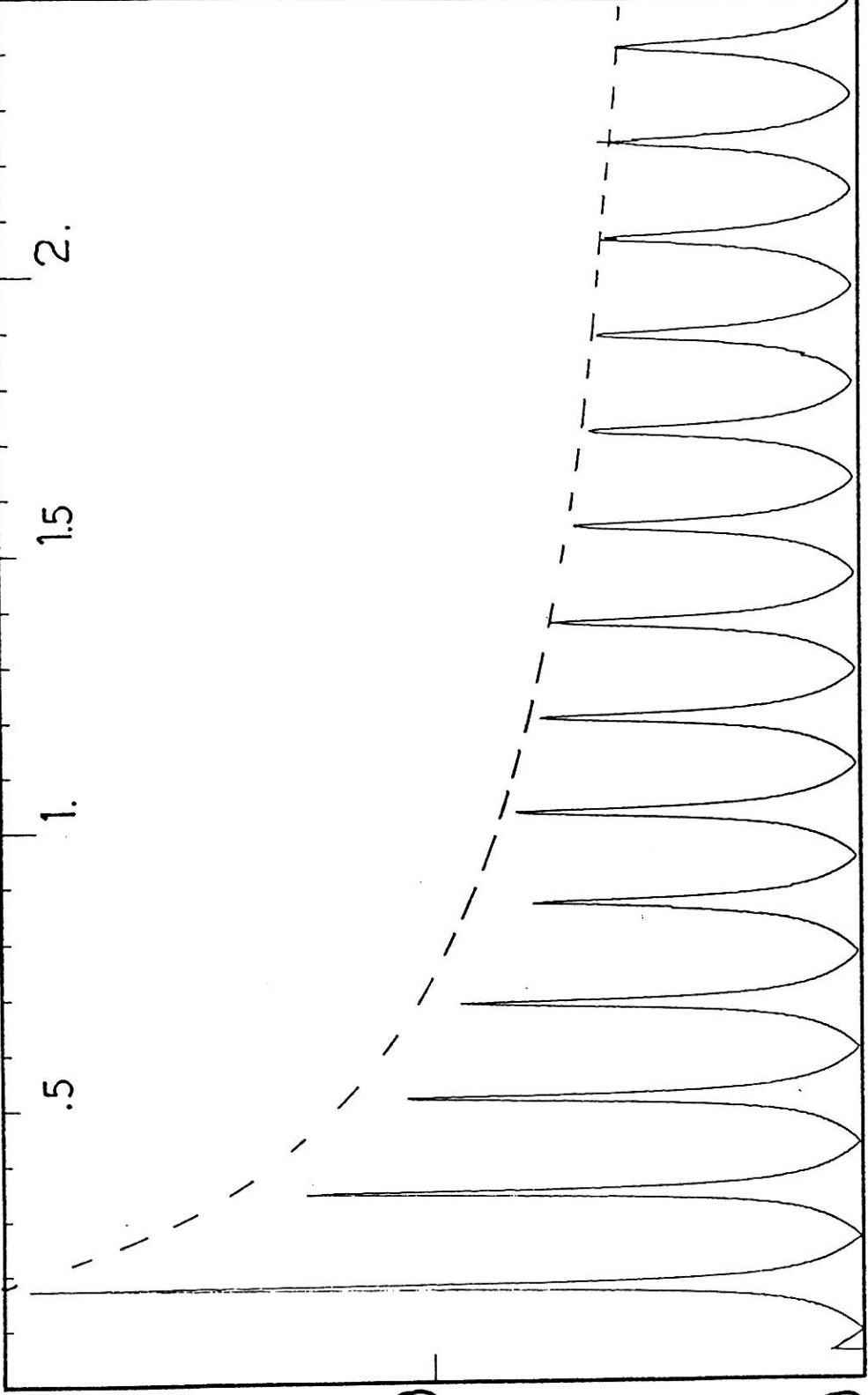


Fig 7

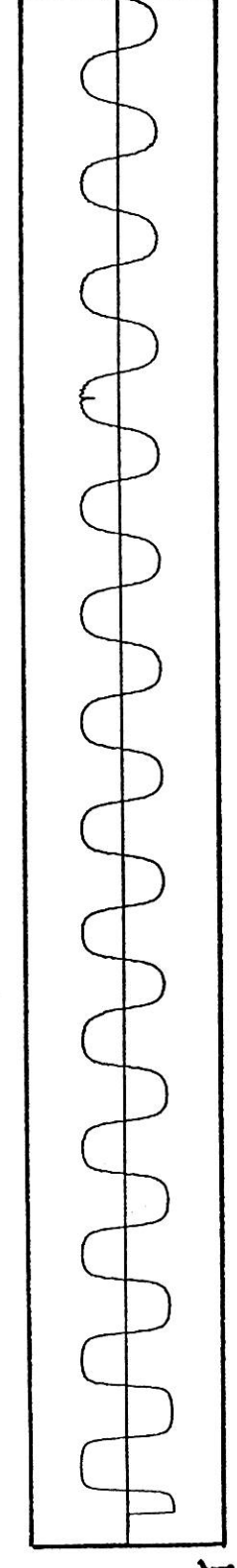


$f_{B,f}$

f (kHz)

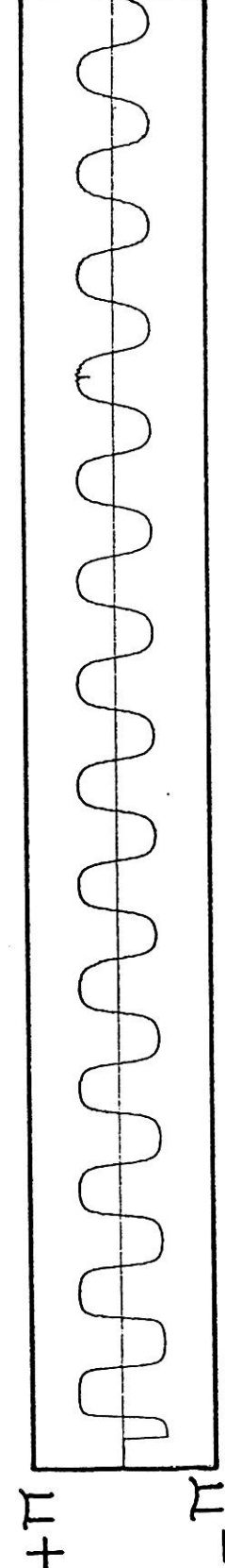
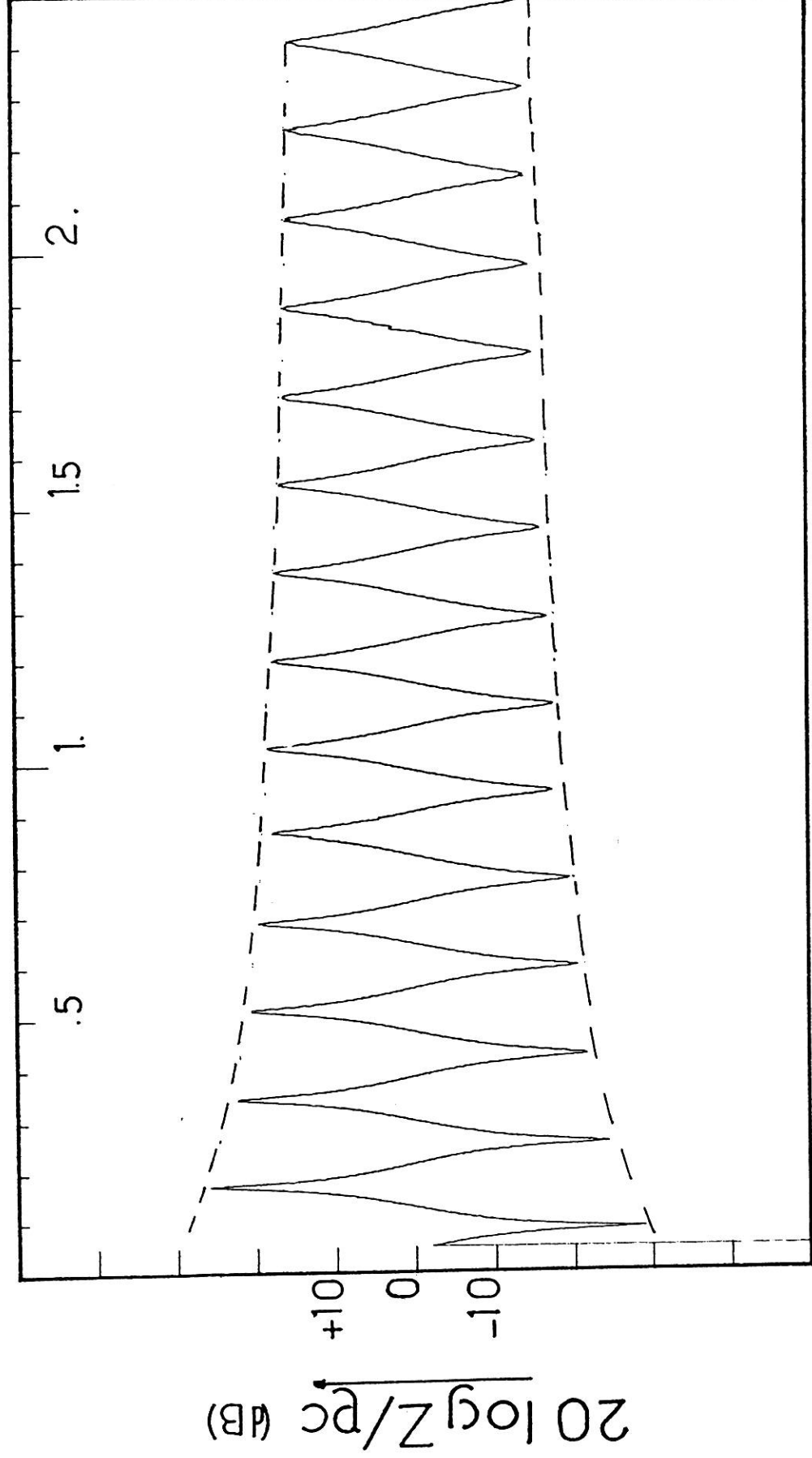


Z/pc



0.9 a

f (kHz)



f (kHz)

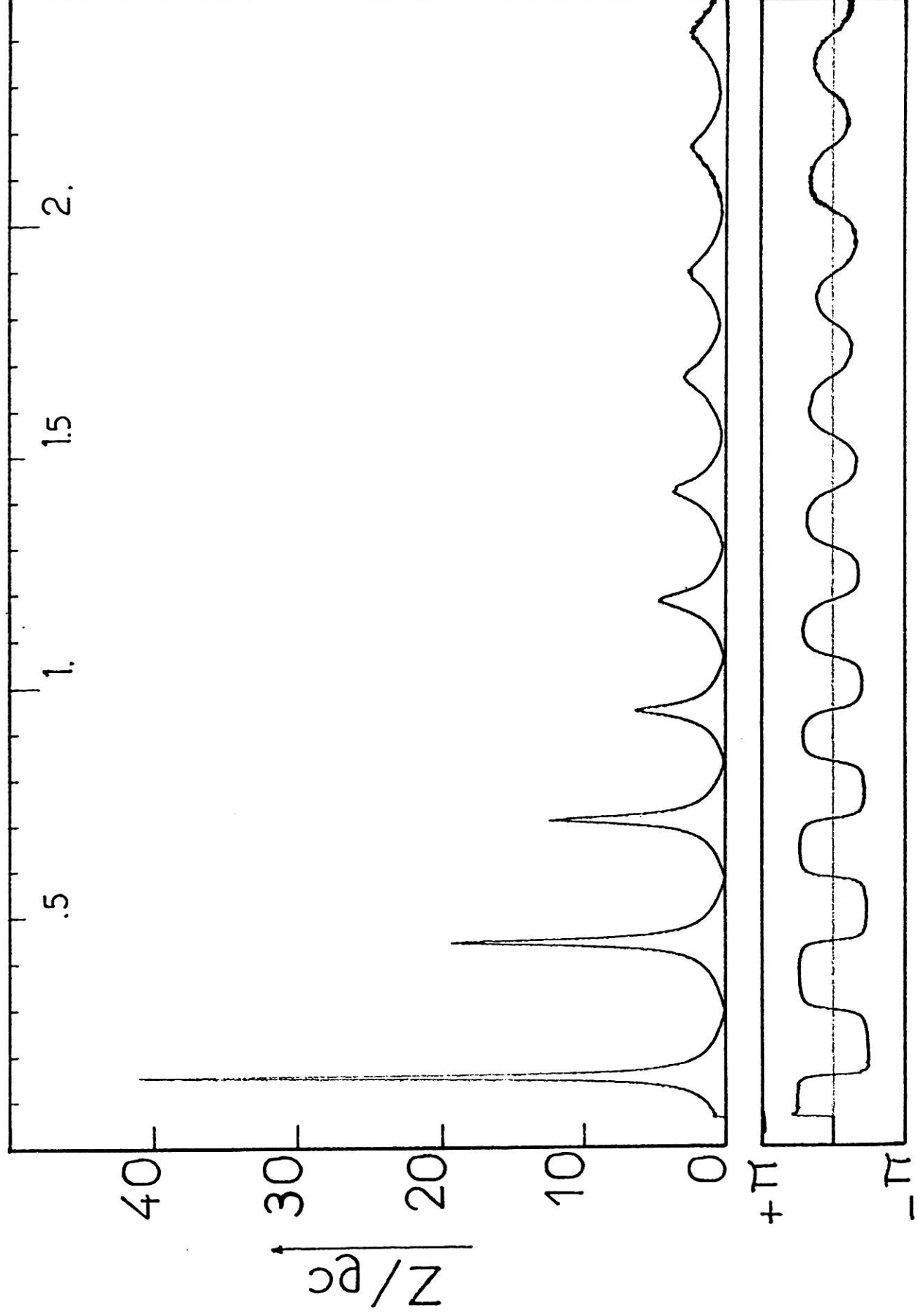


fig 10

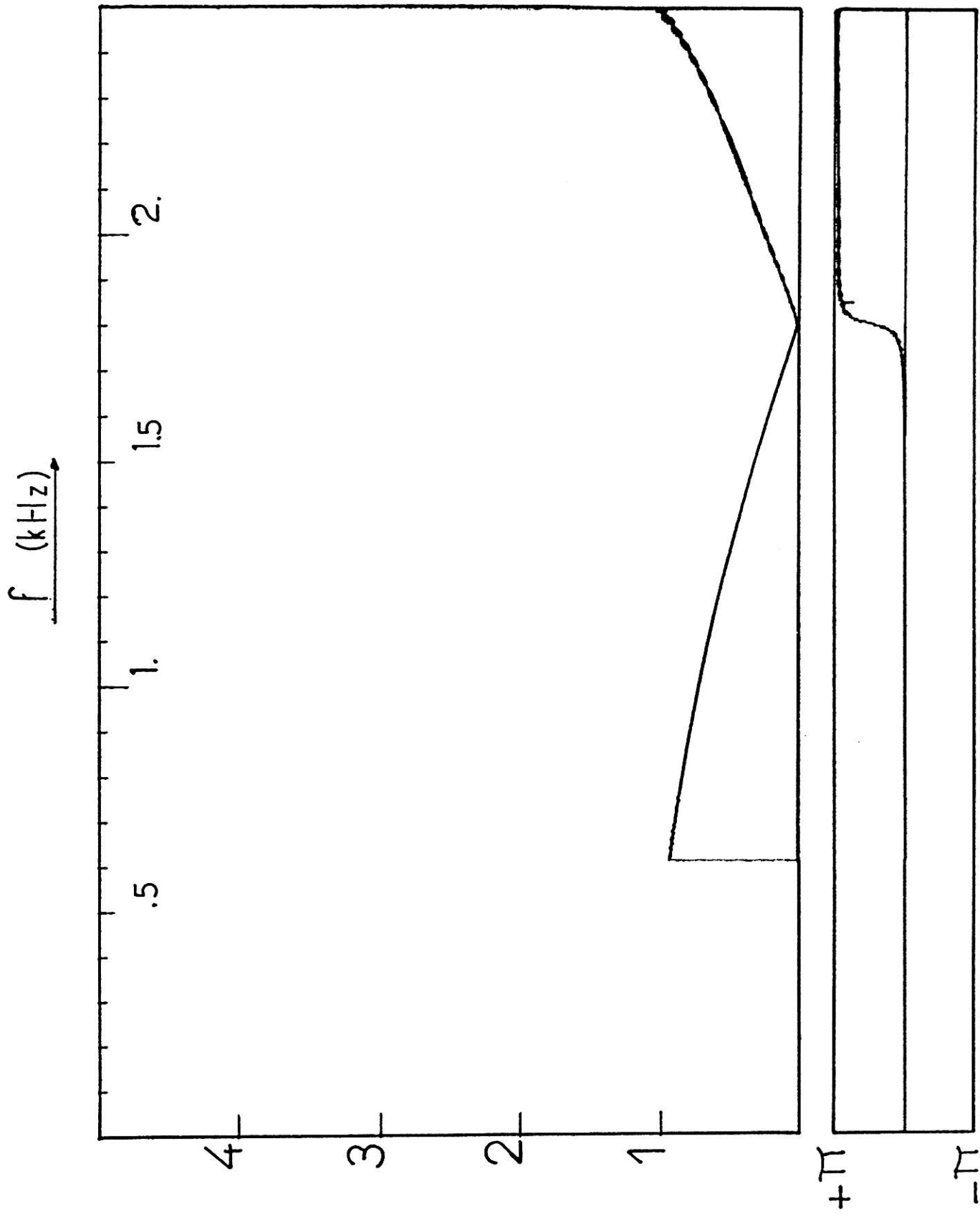


fig 11

F (kHz)

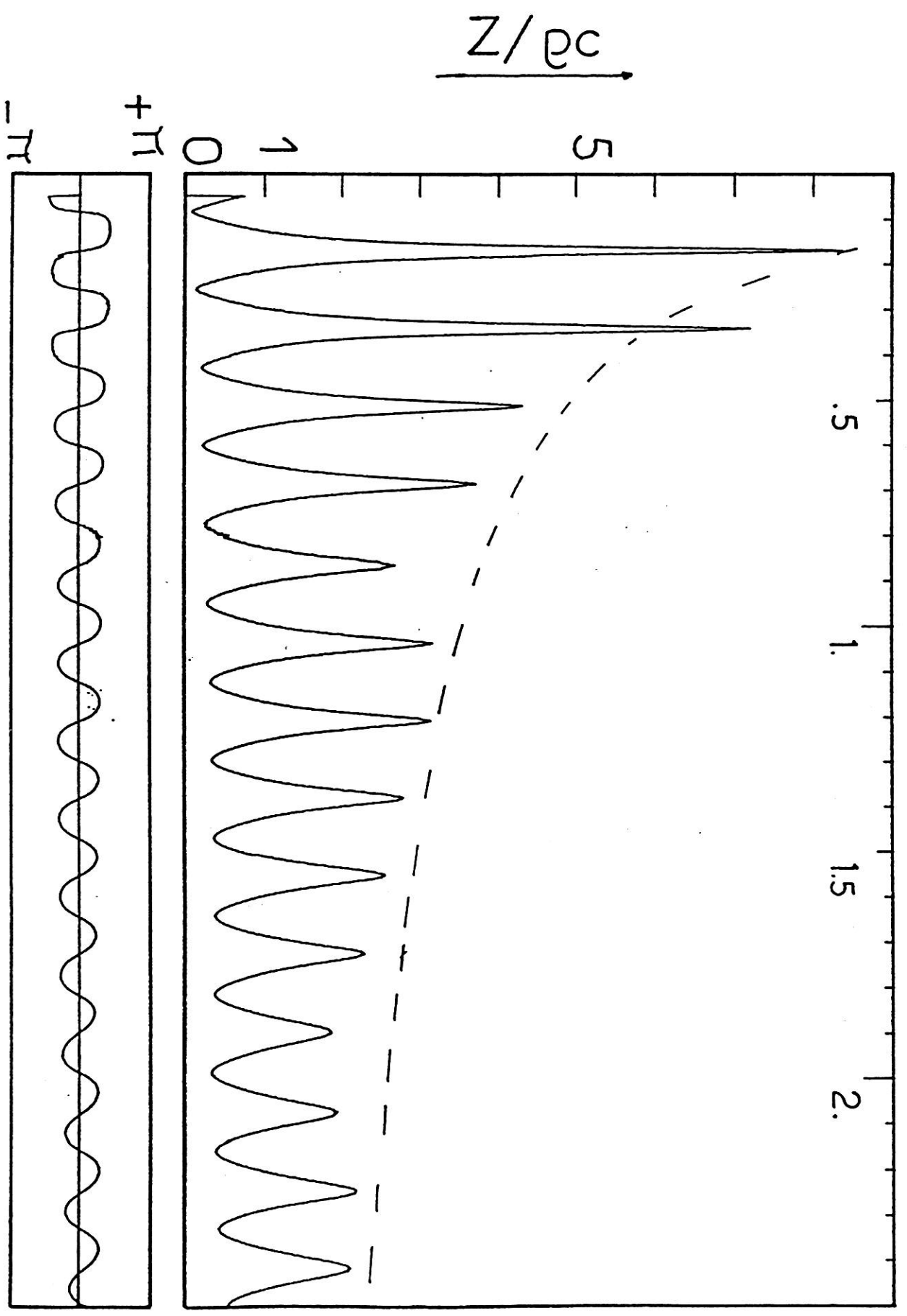


Fig. 18a

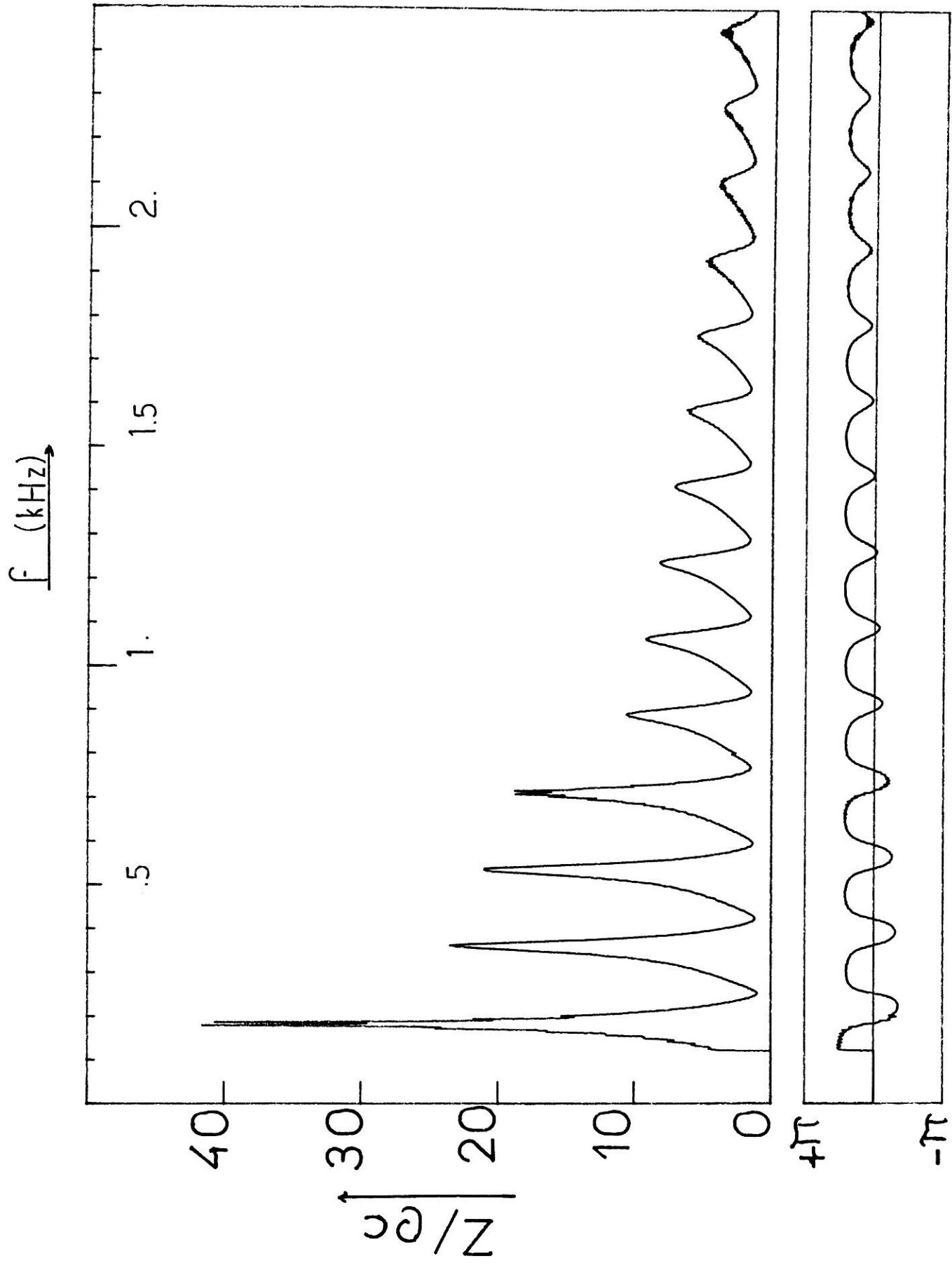
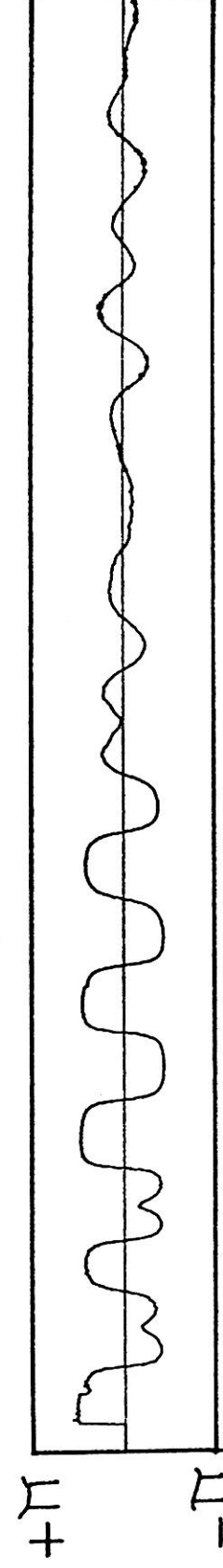
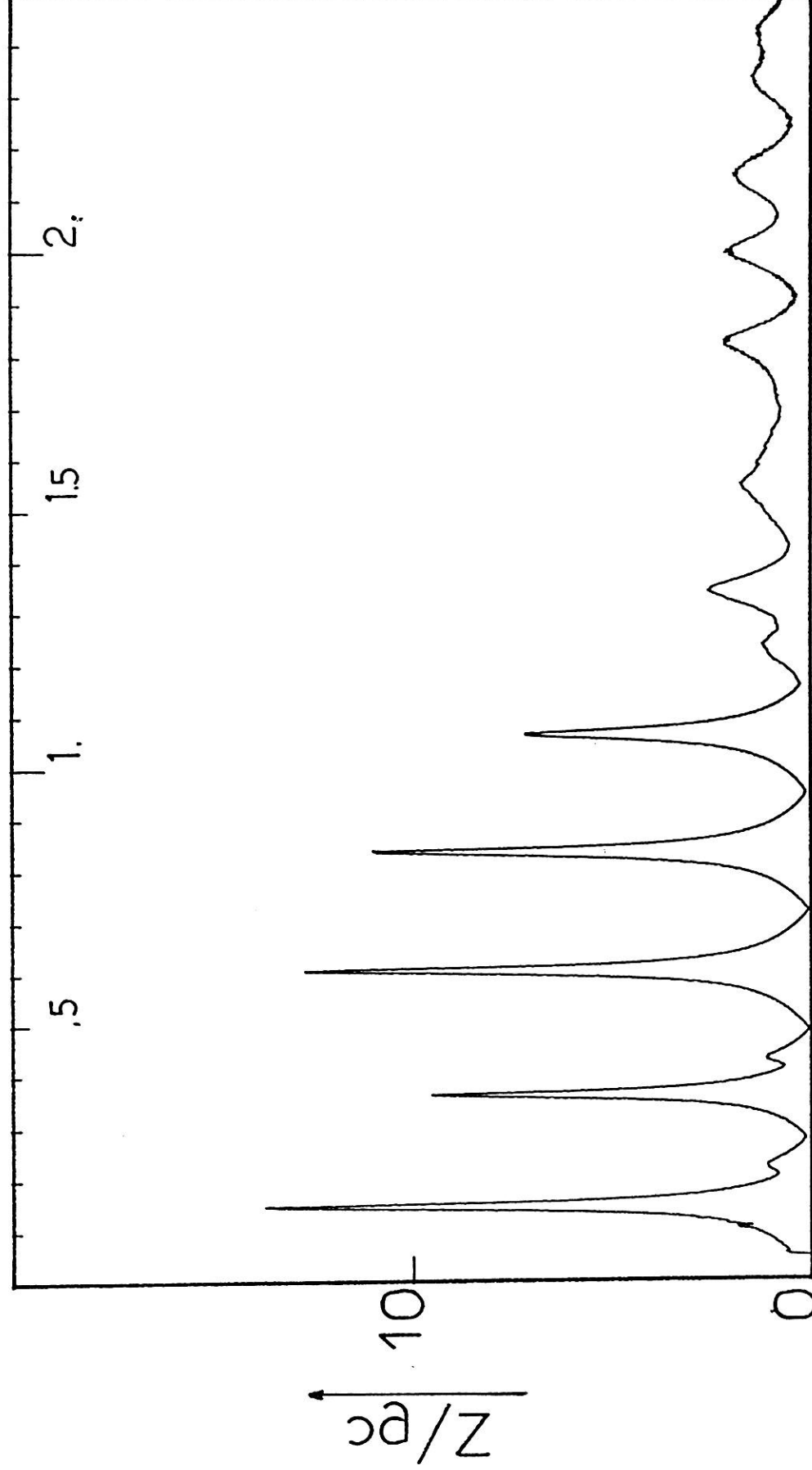


Fig 12b

f (kHz) \rightarrow



$f_{13} B$

Fig 16

