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EXPERIENCE WITH A VERSATILE NLA SONAR SYSTEM

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1. INTRODUCTION

During the last few years the Sonar Signal Processing Group at Loughborough University has developed two, experimental, high power, phased array systems for non-linear acoustic experiments. These employ piston transducer elements to transmit the primary signals with centre frequencies of 40 kHz and 20 kHz. The 40 kHz array was configured from 256 wideband elements connected into 16 parallel staves each of which was driven by a linear amplifier rated at 1 kW. With precision control of both amplitude and phase in each stave it is practical to steer the acoustic emissions dynamically, encode a variety of modulation options and shade the transducer responses to achieve optimum performance in terms of beamwidth and side lobes. The synthesis of the necessary drive signals in the prototype system utilised a simple Z80 based microcomputer to compute 4 bit resolution waveforms off-line. The current system now employs a transputer to generate 16 bit resolution waveforms at speed. Recomputation of complex sets of drive signals now takes only seconds, i.e. almost within a typical long range inter-pulse transmission period.
Both the 40 and 20 kHz systems have been used as non-linear excitation sources to generate low frequency signals in the water and a long range sector scan transmitter employing NLA bearing coded transmissions in conjunction with a single channel receiver/decoder has been demonstrated to be practical.
This paper discusses some of the problems encountered when designing and implementing a versatile parametric sonar.

2. PARAMETRIC SONARS

The non linear (parametric) sonar offers a number of potential advantages in low frequency sonar research which may be summarised as:-

1] High directivity at low frequencies using relatively small transducers.

2] The beamwidth may held reasonably constant over possibly four octaves.

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3] The beam pattern can be made relatively side-lobe free. This factor makes the sonar a valuable tool in reverberation environments.

4] The overall system has a multi-frequency capability with high and low frequency components and their associated harmonics which may find application.

5] The sonar provides a broadband capability in the low frequency regime.

In the low frequency range below about 5 kHz high directivity on reception can now be easily achieved in one plane by the use of long line arrays. To obtain even relatively coarse beam widths on transmission however demands arrays that with conventional systems (in the non military field at least) become completely impractical. At 4 kHz for example, with a wavelength in sea water of 0.375m, a 4° x 4° beam would require a transducer some 16 lambda x 16 lambda or 6m x 6m in linear dimensions: at one kHz the array size would then be 24m x 24m. Using a parametric array with a step down ratio of 10, implying for the former case a carrier frequency of 40 kHz however, this directivity can be achieved with an array of 1/10th of the linear dimensions or 1/100th of the array area. It should be noted here that even this size of array is relatively large and heavy for research purposes.

The original analysis of non linear interaction given by Westervelt [1 & 2] for parametric sources considered only the difference frequency signal generated from plane, narrow, collimated, primary waves. In this case the transducer is driven at two frequencies F_1 and F_2 to establish a high frequency finite amplitude carrier wave in the water. In conventional linear acoustics the signal propagates without modification other than the linear effects of spreading, absorption and scattering. As water however is not a totally incompressible medium, for a finite amplitude wave, the velocity changes in the wave front caused by pressure changes result in considerable wave distortion. At some distance from the sinusoidal source, at high acoustic intensities, a well developed saw tooth wave will be produced. As the wave continues to propagate the harmonic components are absorbed at a higher rate than the carrier frequency and finally, the carrier at long ranges will revert to a sinusoidal form.

This inherent non linearity of the transmission medium also produces intermodulation products to give the sum and difference frequencies $F_1 + F_2$ and $F_1 - F_2$. At the difference frequency F_D

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this results in the generation of a virtual endfire array of sources, generated in a direction normal to the face of the transducer, with approximately exponential amplitude shading as a result of the absorption of the primary wave components. This then forms a narrow beam virtually free of sidelobe structure. It is this narrow transmitted beam, that can be considered, at least to the first order, to have a directivity at the difference frequency $F_1 - F_2 = F_D$ approaching that of the carriers, that is attractive to the sonar engineer. This high directivity however has to be bought at the considerable cost of conversion efficiency which for a 10:1 step down ratio is of the order of 1% or -40 dB with respect to the level of one of the individual carrier signals.

Four distinct operating conditions pertain in general to parametric arrays:

At low transmit levels the primary high frequency wave is limited by small signal absorption. The low frequency beamwidth is then independent of the signal input and the secondary source level is proportional to the square of the primary input.

The virtual array length can be determined in either the near field or in the far field of the primary beam producing two distinct operating characteristics. Westervelt analysis [1] therefore strictly refers to the first of these conditions (i.e. low amplitude signals collimated by the near field of the array).

As the transmitted power is increased the parametric source length becomes limited by non linear absorption in the near field as harmonic distortion and shock formation take place. In this mode of operation the secondary beam pattern broadens as the virtual array length decreases. The parametric gain (the difference in Source Levels between the difference frequency and the primary frequencies) becomes independent of the primary Source Level.

The final case of saturation limiting in the far field is probably of little practical importance as small signal absorption generally limits virtual array length.

This paper describes the development of two parametric sonars specifically designed to generate low frequency signals in the 0.5 to 5 kHz spectrum. As will become apparent from their design features both these sonars are of the type that are absorption limited in the far field.

3. SONAR DESIGN CONSIDERATIONS

The definition of a parametric sonar in terms of SL_D(Source Level at the difference frequency) and beam width is determined in a relatively complex manner by the parameters of the primary source and the step down ratio involved. The main controlling factors are the form, frequency and the Source Levels (SL_1 & SL_2) of the primary carrier signals, the transducer aperture and the small signal absorption coefficient (α in neper/m). In the case of a system where the majority of the difference frequency is generated in the near field SL_D becomes inversely proportional to the 4th power of the step down ratio. When the majority of the difference frequency is generated in the far field SL_D becomes inversely proportional to the square of the step down ratio. For high efficiency of operation therefore it is preferable to preserve a small step down ratio and operate the sonar as a far field source.

SL_D is also proportional to the sum of the Source Levels of the two high frequency carrier signals $SL_1 + SL_2$ (generally virtually equal) but as has been noted, at high primary Source Levels further increase in SL_D is limited by saturation effects. For high levels of SL_D high levels of SL_1 and SL_2 are required while preferably avoiding a situation of uneconomic transmitter power requirements by driving the system into saturation.

For scientific research purposes it is obvious that, in general, the sonar should be used at ranges greater than that of the virtual array length. It is only in the parametric array far field that the full source level is generated and the ideal beamwidth obtained (for simple detection purposes these particular limitations may be of no great importance). This consideration does create some problems in system design for to preserve small step down ratios the mean carrier frequency must be well below 100 kHz. The virtual array length is conventionally (and approximately) defined by the term $(2\alpha)^{-1}$ because the source density is attenuated as $e^{-2\alpha r}$, where r is the distance from the primary projector. In fact the effective length of an absorption limited array can vary from about 10 x $(2\alpha)^{-1}$ to half $(2\alpha)^{-1}$ depending on whether the majority of the frequency generation takes place in the near field or the far field. In the cases considered here the term $(2\alpha)^{-1}$ gives a good enough approximation to illustrate the present problems. Figure 1 shows the factor $(2\alpha)^{-1}$ plotted for frequencies 15 kHz to 80 kHz. At 20 kHz for example the virtual array length is nearly 1500m whereas at 80 kHz it is less than 150m.

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If the system is to be used as a calibrated source then working within the virtual array length has to be approached with considerable caution. Not only will the SL_D be reduced at shorter ranges but the beam width will be broadened. In addition to these factors at short ranges the effect of the carrier signal itself has to be taken into consideration. In the laboratory this problem can be eliminated by truncating the array by means of an acoustic filter designed to pass only the low frequency component. In the sea this option is not available and the component of the carrier signal reflected from a large target may be sufficient to cause significant secondary regeneration on the return path. This could cause obvious problems in making absolute measurements of the level of back scattered energy.

In system design therefore a compromise has to be made between the need for low step down ratios and the desirability of a relatively short virtual array length.

It is of major importance for an experimental research sonar that the array is transportable and able to be reasonably easily installed in what are generally temporary on-site facilities. This to some considerable extent set an approximate limit to the transducer size and design options based on a 0.6m x 0.6m transducer surface area were therefore examined.

As has been noted, the SL_D at the difference frequency decreases as the step down ratio increases and for this type of array it is of the order of 10dBs for a halving of the difference frequency. As we reduce the carrier frequency however the Directivity Index (DI) is also reduced and for the same available power a halving of the operating frequency will reduce the DI and therefore the Source Level of each carrier frequency by 6 dB. As SL_D is proportional to $SL_1 + SL_2$, SL_D will fall by 12 dB.

This overall loss in SL_D is more than compensated for by the reduction in absorption at the lower frequency and the increase in wavelength. The increase in SL_D however is relatively small unless the available power can be increased to compensate for the loss in Directivity Index and thus maintain $SL_1 + SL_2$.

For carrier frequencies 20 kHz to 50 kHz it was assumed that the array would be constructed of piston elements of size about 1λ by 1λ each of which could handle some 300 W of electrical power. At 40 kHz therefore there would be 256 elements capable of handling 76.8 kW of electrical power but at 20 kHz there would only be 1/4 of that number giving a power limit of 19.2 kW. If consideration is given to carrier frequencies below 20 kHz the individual elements would probably be of a type similar to the ITC-3001 which will give a 50% efficiency to a 2 kW input. In this case however individual transducer size reduces the number of elements in the array to about 9 providing for an electrical limit of 18 kW. (Note - the use of large array elements would severely limit the application of scanning techniques in this system.)

A second limitation at high powers in piston type design elements results from restrictions placed on the transducer by the available thermal dissipation through the array itself and the ability for cooling through the array 'window'. A CW limit of some 2 W/cm^2 is set by this factor which in the present case would set a limit of 7.2 kW for this array. For most pulse lengths and duty cycles this may not provide the ultimate limiting consideration but it is one which needs careful attention during array design and sets a premium on transducer conversion efficiency.

If we lower our carrier frequency too far we are likely to run into problems due to cavitation. Figure 2 shows the predicted cavitation limits in terms of W/cm^2 at the transducer face for shallow water working and for operation at 30m depth. For the later case two curves are shown, the lower one corresponding to a very conservative limit for cavitation onset, whereas it is probable that under practical conditions the much greater level indicated by the higher curve can be transmitted.

For near surface operation the loading is restricted to some 10W/cm^2 at 80 kHz, reducing to less than 1W/cm^2 at 10 kHz. At 30m depth these figure become nearly 20W/cm^2 and 6W/cm^2 respectively. With the transducer surface area specified as 3600cm^2 the acoustic power radiated becomes 36kW at 80kHz and only 3.6kW at 10kHz for near surface operation. At 30m depth it would appear possible to radiate about 72kW at 80kHz and probably nearly 22kW at 10kHz.

At high intensities the waveform becomes distorted, harmonics are generated and finally a 'shock front' may develop. Until a shock front is formed the additional energy dissipated by these factors is relatively small but at longer ranges the waveform becomes considerably attenuated reducing the fundamental component.

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For seawater the shock front has been shown to occur at the Rayleigh distance R_0 given by A/λ (A= transducer surface area) if the rms Source Level is given by:-

$$SL_0 = 280.5 - 20\log(F_0) \quad \text{dB re 1 } \mu\text{Pa at 1 m}$$

Where F_0 is the carrier frequency in kHz.

This give rise to the concept of a 'scaled up' Source Level SL^*

$$SL^* = SL_0 + 20\log(F_0)$$

Only if SL^* is well below the 280.5 dB level will the spherical wave region be reached without excessive distortion and attenuation. In order to be reasonably economical with the requirements of the power amplifier system and to avoid saturation effects it is desirable to work below these saturation limits. Berktay [3] suggests maximum Source Levels useable for non saturated working. These levels are somewhat different from those defined by considering the criterion of shock front formation at R_0 alone.

Finally we have a limitation defined by the practical conditions of available power. A power amplifier system capable of driving an 80kHz transducer of this size to cavitation at 30m depth would probably have to be considered completely impractical - even if there were worthwhile acoustic dividends to be achieved.

In Figure 3 these various constraints are shown as a function of frequency for a transducer of the size considered, i.e. 0.6m x 0.6m, for various Source Levels. Shown also is the SL_0 available for an electrical power level of 16 kW assuming transducer efficiencies of 50% giving an acoustic output of 8kW into the water for single frequency transmission.

As can be seen with this level of available power below about 25 kHz, operation would be well below any effects of saturation but would be verging on the cavitation limit even at 30m depth.

Above 25kHz the sonar would be well within the cavitation limit but as the frequency is raised it would be operating closer and closer to a saturated array situation with the inherent problems of beam broadening and loss of transmission efficiency.

Consideration of all these design aspects led in the first instance to the definition of a sonar based on a 0.6m x 0.6m transducer operating at 40 kHz to provide a 16λ x 16λ array with beamwidths of approximately 4° x 4°.

Operation of such a single channel system with relatively narrow beamwidths is always constrained by the low search rate involved and the difficulties of beam positioning. In order to reduce such problems and to provide a much more versatile system it was decided to construct a scanning transmitter. With a 16λ x 16λ array an appropriate system is one of 16 channels providing for scanning over a sector of 60° in one plane (azimuth). An economic power drive arrangement was therefore of 16 x 1kW power amplifiers each matched to drive one stave of 16 elements. This provides for a single frequency acoustic power of 8kW or in the two frequency drive case two signals each of 4kW. Such a transducer at 40kHz has a DI of 35dB and a near field R_0 of 9.6m for a single CW signal the calculated SL_0 is therefore 244.8 dB re 1 μPa at 1m.

Although reduction of the carrier frequency to frequencies lower than 40 kHz causes some practical problems, particularly for short range working, the success of the original 40kHz sonar has led to the further exploitation of this parametric technique using a 20kHz carrier frequency. In this case a transducer of identical dimensions to the 40 kHz array has been developed with an array of 8 x 8 elements. In this array all elements are individually accessible permitting, in the ultimate, the development of scanning in both azimuth and elevation.

4. TRANSMITTER DRIVE WAVEFORMS

There are several methods by which a non linear sonar may be implemented. In the original form due to Weltervelt two individual signals of frequencies F_1 and F_2 are linearly added and applied to the transducer. This two frequency system is frequently termed the Double Side Band Suppressed Carrier (DSSC) mode of operation and has been analysed in depth by many authors.

This form of drive function does however suffer from a number of problems. Foremost amongst these probably is the fact that the two frequencies exist in the power amplifier drive system which must therefore be ultra linear. Any non linearity will cause the generation of a frequency component $F_1 - F_2$ in the power amplifier itself which will then be radiated only with the directivity associated with the difference frequency. This can negate the advantages sought in the high directivity associated with the parametric sonar itself. To eliminate this effect transducers have been subdivided in chess board fashion (with associated loss

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of DI) or in more extreme cases two individual transducers have been used, one for each frequency.

A second problem is the fact that by adding the two signals linearly the phenomenon of 'beats' occurs and the resulting peak voltage amplitude becomes twice that of the individual signals. This requires at maximum amplitude 4 times the peak power of each signal individually. If the power amplifier system is peak power limited as is frequently the case the levels of SL_1 and SL_2 have to be appropriately reduced. The inherent low conversion efficiency of the non linear generation has prompted investigations into how source levels may improved. Merklinger [4 & 5] has shown that in principle it should be possible to obtain 4 times as much average power at the difference frequency by transmitting the same average power in a different fashion. Practically the optimum solution cannot be realised, however if a single frequency carrier is amplitude modulated by a periodic function it can be shown that the overall efficiency can be improved.

The basic CW pulsed signal can be modulated for example by a sinewave function giving rise to a signal at F_0 and sideband signals at F_0-F_D and F_0+F_D. The two sidebands will then interact non linearly with carrier during transmission through the sea producing a signal at F_D. A signal at lower level will also be produced at $2F_D$ due to the interaction of the sidebands themselves.

If the carrier is modulated with a square wave with a 1:1 on/off ratio, many more sidebands are generated with a theoretical improvement in SL_D of 2.1 dBs. In this form of modulation (AM mode of working) the peak envelope power is twice the average power (due to the 1:1 switching ratio) which is exactly the same as that of DSSC operation. If the primary waves are of very high intensity the low frequency signal become linear with input rather than square law dependent and the AM method of working will provide an improvement in efficiency of 3.5 dB. Confirmation of these predictions has for example been made by Carlton [6] who obtained Source Levels up to 2 dB higher using the AM mode when compared with DSSC operation.

The AM system has a second advantage over the DSSC mode in the fact that it is possible to use more efficient, very much smaller, cheaper switch mode power amplifiers.

The principle disadvantage of AM working is that it requires double the bandwidth in both transmitter amplifiers and transducer compared with DSSC systems.

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The difference in the spectral components detected in the water for the two types of transmission are shown in Figures 4. For the two frequency system in this case transmission is made by two carrier frequencies of 44kHz and 41kHz to provide a difference frequency of 3kHz. Together with these two primary signal components the received spectrum at 300m includes the intermodulation products generated by the transmission medium. At around 80kHz are the second harmonic components together with $F_1 + F_2$ and at 3 kHz is the required difference frequency.

For the square wave modulation, Figure 5, we have the primary components at F_0, F_0-F_D and F_0+F_D together with the series of components generated by the square wave modulation function. At the low frequency end of the spectrum we have the wanted 3kHz component together with a series of additional low frequency signals at 6kHz, 9kHz, 12kHz etc. whose level approaches that of the wanted signal. This multiplicity of low frequency signals can, in some cases, be made use of but may also limit the available bandwidth that can be used at the low frequency end of the spectrum.

In this case the improvement in Source Level for AM operation as opposed to DSSC working was found to be 1.22dB.

5. TRANSDUCER DESIGN

Wood et al. [7] describes the development stages which led to the chosen configuration of the 16λ x 16λ 40kHz array. Primarily due to a manufacturing (machine capacity) limitation it was decided to implement the array construction in two halves. This decision had benefits which became apparent when the performance was computer modelled. The split array enforced a discontinuity in the element spacing across the join and it was found that the deliberate introduction of a half λ spacing significantly reduced the amplitude of the unscanned (vertical) endfire component. In addition the two array halves could be separated for independent use and the first half was commissioned and fully tested before the second was assembled.

The array construction employed a pocketed nylotron housing with the wideband 40 kHz elements supported on pads of syntactic urethane foam. The element assembly is sealed in place with a thin but very tough ρc Adiprene window which bonds to the faces of each piston and provides the thermal and acoustic coupling path to the water. The rear of the array housing incorporates a junction box, behind the element support bulkhead, in which the electrical connections are made to waterproof cable sockets. Each group of 4 elements (substave) was connected in parallel to its individual drive signal. At the cable (surface) end these connections were paired together to make 8 element (half stave) groups for driving by a channel amplifier output. The completed 256 element transducer array was therefore connected by two umbilical cables each comprising bundles of coaxial inner connections which carry the power amplified drive signals for each stave. Even with 8 elements grouped together in parallel the load impedance is relatively high and peak voltages in the order of 1.1kV are needed to achieve the desired Source Levels. Careful consideration of the peak voltage stress is required when selecting the connectors and cables. A marginal specification may survive (dry) laboratory proof tests but corona discharge damage can occur if waterproof connectors are made up in high humidity conditions!

6. SIGNAL SYNTHESIS HARDWARE

The versatility of any phased array sonar largely depends on the precision with which the drive signal amplitude and phase can be defined in each transducer stave and the ease with which these waveforms can be modified. The application of a microcomputer to generate matched sets of sonar signal waveforms digitally has been described by Goodson [8], Goodson et al.[9] and by Cook et al. [10,11 & 12]. Digitally synthesised techniques applied to the 40 kHz 16λ x 16λ phased array can provide signals for:-

i] Wide band primary signal frequency synthesis, limited only by the transducer bandwidth, including frequency chirps of almost any desired complexity.

ii] Steering of the primary transmission direction with respect to the array axis to any angle within the range +/-30°.

iii] Ripple-fire transmissions - a contiguous stepped transmission across a selected sector.

iv] Sweep transmissions - a continuously swept bearing transmission through a selected sector at a controlled sweep rate.

v] DSSC and AM Modulation (applicable to all the above) to generate steered NLA signals. Low frequency signal generation from 0.5 to 6kHz and high frequencies at 2 x F_0.

vi] Amplitude shading of the pulse envelopes and of the drive levels to implement shading. Dolph-Chebychev and Taylor weightings have been used to control sidelobe/beamwidths in primary frequency applications.

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vii] Focussing - progressively advancing (or retarding) the phase of the outer elements provides a focussing effect which can usefully alter the near/far field transition of the primary frequency transmissions.

viii] A variety of test signals for channel matching (phase & amplitude) and system alignment.

ix] Complex signal synthesis - coded concurrent multibeam signal generation.

The prototype Z80 based system described in [8] has now been superseded by a more versatile synthesiser based on a single INMOS T800 Transputer [13 & 14], figure 6. The operator interface and system control handled by an IBM AT computer. This combination gives the sonar operator menu access to a variety of signal parameters and, after selection, these are transferred to the Transputer signal generator. The computation of a complete set of wave forms, needed to drive all the staves of the transducer array, occurs sequentially and each synthesised digital waveform is stored in a RAM memory buffer assigned to each channel, Figure 7.

Once filled, the common data bus is disabled and control of these memory buffers is reconfigured to address them all simultaneously. The data output from each buffer feeds one of 16 D/A converters. These signals, with minimal filtering, produce the multiple, phase related, analogue signals used as drive voltages by the power amplifiers.

The transmitted signal duration, especially where coding demands a unique signal, is primarily determined by the available output buffer length and by the sampling frequency. The original Z80 was configured to page address, with some mapping restrictions, 8 off 32 kilobyte buffers, each buffer storing 2 multiplexed 4 bit waveforms. The current transputer system is less restrictive and provides a 64k x 16 bit memory buffer for each channel. Sampling frequencies are normally set at 250 kHz, much higher than the Nyquist limit for the highest frequency generated, as by oversampling very wideband signals can be generated with a minimum of output filtering.

The high power linear amplifiers [9] used to drive each stage of the array are a modification of a commercial design using MOSFET power transistors and each is capable of producing over 1 kW under near CW conditions. Each power amplifier output drives the primary of a matching transformer which employs matched dual secondary windings to drive the two halves of the array.

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7. RECEIVERS

The basic parametric sonar will only provide high directivity on transmission although parametric receivers using similar principles have been developed [15]. Whilst it may be possible to use the transmitting transducer itself for reception in some circumstances, an NLA transmitter preferably requires a dedicated separate receiver. In the present case the scanning transmitter insonifies a sector of 60° (horizontal) x 4° (vertical) and the receiving array should ideally be matched to this volume. Unfortunately, whilst the transmitter operates at a relatively fixed carrier frequency, the receiver may be required to operate over several octaves from say 500 Hz to 6 kHz and to provide even an approximate match over this entire range is not practical. For the present system a low frequency line array has been used as a receiver. This consists of two vertical arrays, each of ten elements at 0.3m spacings. At 5 kHz this provides a 1λ (60°) x 9λ (6°) array and at this frequency approximately matches system requirements.

A single channel receiver however, has no direct means available to identify from what part of the scanned sector echoes are being received but by individually coding the transmission on each bearing the incoming signals returning from a given direction can be positively identified in a receiver processor. Many sets of orthogonal codes exist, one of the more interesting being described by Jaffe & Cassereau [16]. The major consideration is for a narrow autocorrelation function to provide range resolution and a small cross correlation function to ensure rejection of adjacent channels. For initial development a very basic code has been chosen providing no autocorrelation but reasonably low cross correlation between channels. This is simply a set of frequencies each of which differs by more than one bandwidth of the transmitted signal. The receiver can then be a set of filters each tuned to the appropriate frequency. At 40 kHz a bandwidth in excess of 10 kHz is available but to prevent difference frequency harmonic interference it is necessary to limit the available bandwidth to less than one octave at F_D. A 10ms pulse with an associated bandwidth of 100 Hz in the present case limits the total bandwidth required to 1,600 Hz which provides a practical compromise.

The received signals are first amplified, provided with time varied gain control, and then low pass filtered. In the 40 kHz system the filter used has a 12 dB per octave attenuation characteristic which provides initial compensation for the change in efficiency of the non linear conversion which decreases with decreasing frequency.

The signal is then high pass filtered to define the wanted pass band, buffered and fed to a bank of 16 narrow band filters with 3 dB points +/-50 Hz from their centre frequencies. The filters feed individual detectors and the resulting 16 output signals are multiplexed into 8 bit A/D converters and displayed visually via a frame store. The data can then be displayed sector corrected or in B-scan format. A newer all digital decoder/display system has also been recently tested. This comprises an IBM 386 SVGA microcomputer hosting a Motorola 56001 DSP card. The preamplified and bandlimited analogue signals are digitised (12bit) and the DSP processor implements a digital filtering algorithm to separate the channel information. A variety of displays are being examined but the pseudo 3D 'waterfall' display as in Fig.12 provides an interesting alternative to the more usual intensity modulated image.

8. SOURCE LEVELS AND BEAM PATTERNS

Experimental measurements were carried out in a deep water Scottish sea loch with the transducer arrays deployed some 30m below the surface. Calibrated long cable hydrophones were used to receive the transmissions. These were placed at the same depth as the transducer but were spaced at intervals out to 500m down range.

Measurements of the pressure levels of the various frequency components have been made at a number of ranges using a wide bandwidth passive hydrophone and a frequency analyser. The linearity of the transmitter level was first examined by plotting the single carrier frequency source level, as measured by a hydrophone at 90m range, against the number of active staves in the 40 kHz array. In figure 8 the non-linearity becomes apparent as the 40 kHz array SL_0 increases with the addition of the outer staves. The Source Level of the 20 kHz array, driven with slightly lower power and with its lower DI does not show this effect. In a totally linear system the Source Level should increase by 6 dB for each doubling of the number of active staves, 3 dB due to the increase in DI and 3 dB due to the increase in power. From the one to sixteen stave case therefore the SL_0 should increase by 24 dB. For a single stave, driven at 42 kHz the SL_0 achieved was 222.43 dB re 1 μPa at 1m. This leads to an expectation that at full power 16 staves should contribute an effective SL_0 of 246.43 dB re 1 μPa at 1m. In practice at 42 kHz the short range SL_0 was measured to be 244 dB re 1 μPa at 1m.

As the transducer array was supported by a computer controlled training gear it was possible to obtain accurate beam plots by plotting the hydrophone response as the transducer was rotated in azimuth in small angular increments. Beam plots of the primary transmissions of both the 40 kHz and 20 kHz arrays are shown in figure 9 together with NLA secondary beam plots at 4 kHz.

In figure 10 actual and predicted (infinite range) values of SL_D are plotted for a range of secondary frequencies. These predictions were based on the measured values of SL_1 & SL_2 (DSSC modulation). The 20 kHz array data flattens at around 186 db re 1 μPa above 4 kHz as the transducers lacked the bandwidth to maintain SL_1 & SL_2. More recent tests, driving the array with the maximum available power (short pulse rating), demonstrated that the 20 kHz array SL_D increased by a further 7.75 dB (DSSC modulation) and by 9 dB when using AM. These drive levels require the transmitter to deliver 12kW acoustic power into the water and it is unrealistic to expect to exceed an SL_0 of 240 dB re 1 μPa from this system.

In general the measured levels for SL_D were found to be in reasonable agreement with those predicted over the complete range of frequencies measured.

Figure 11 shows the echoes of a single ripple-fired transmission scanning a Scottish sea loch, approximately 800m wide and with water depth up to 80m. The hard copy taken from the video display is for a single swept frequency transmission of 2.2 - 3.7 kHz. The equivalent pulse length on each bearing is 30ms and hence there is an uncorrected range distortion from left to right across the display. Transducer depth was 23m and the field of view at ranges in excess of about 1000m was shadowed on the right by a zone of much shallow water which extended into the loch near a headland. On the left clear water exists up to a headland (display centre) at about 3200m. Maximum range displayed is 3700m (left) and the rapidly shelving shore line is recorded up to a maximum range on beam 7 of approximately 3400m.

The results obtained with this non-linear transmitter scanned sonar show good agreement with those predicted from the design parameters and the effectiveness of sector scanning techniques employing simple bearing coded emissions has been demonstrated.

Figure 13 shows the echo detection of a -6 dB TS sphere deployed at 300m range at 3 kHz. The first image using an NLA generated pulse is significantly better than that obtained by a conventional 3 kHz sonar. The higher DI and lack of sidelobes in the NLA transmission accounting for the improved signal/reverberation ratio. The NLA receive array was used for T/R in the linear case.

9. ACKNOWLEDGMENTS

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EXPERIENCE WITH A VERSATILE NLA SONAR SYSTEM

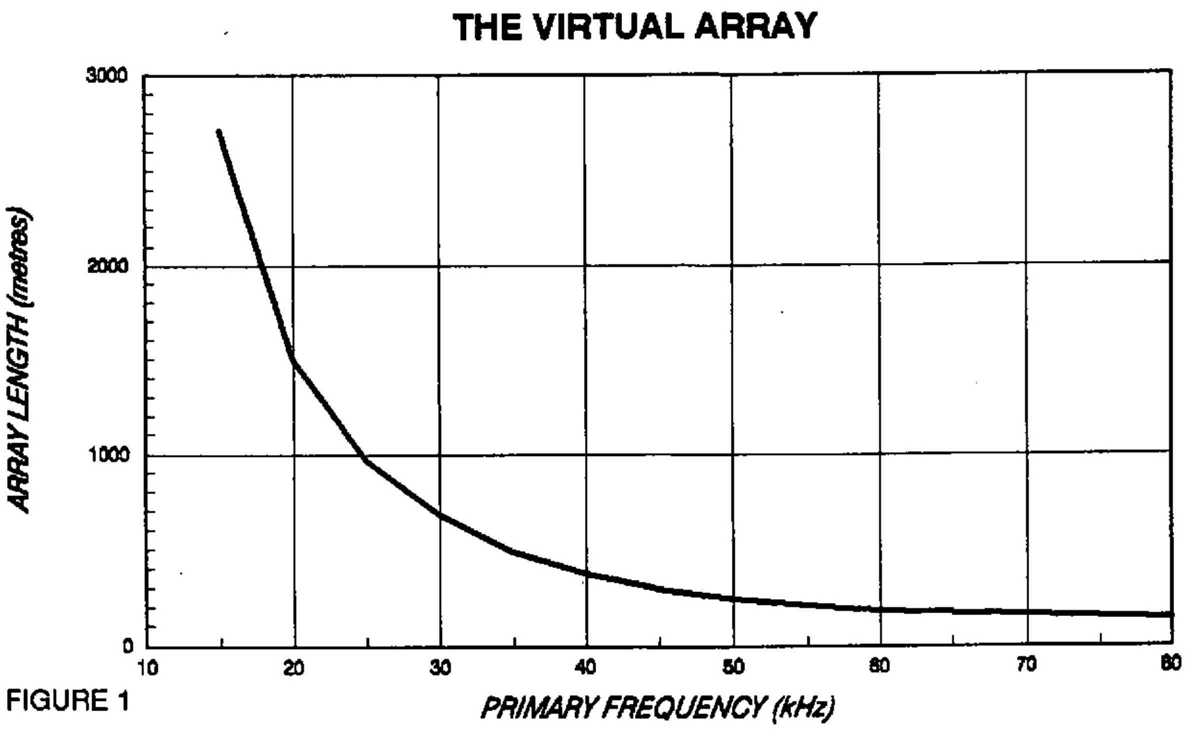


FIGURE 1

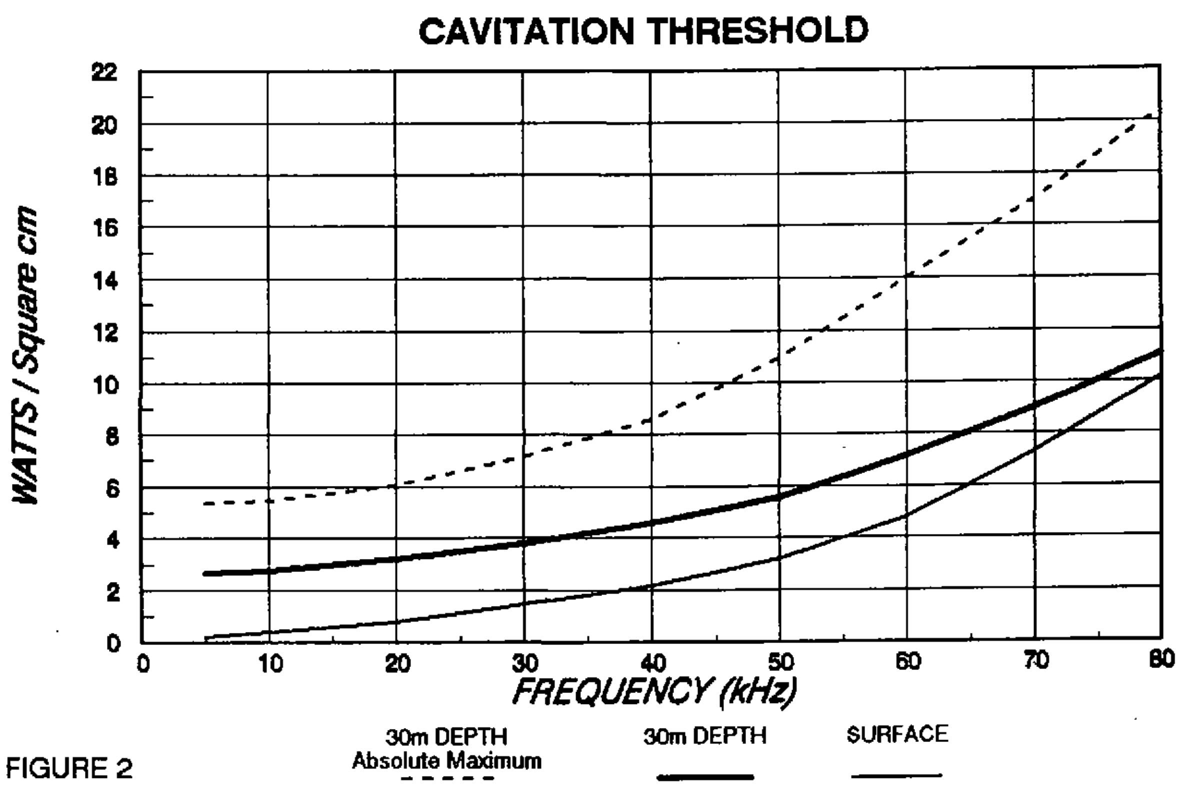


FIGURE 2

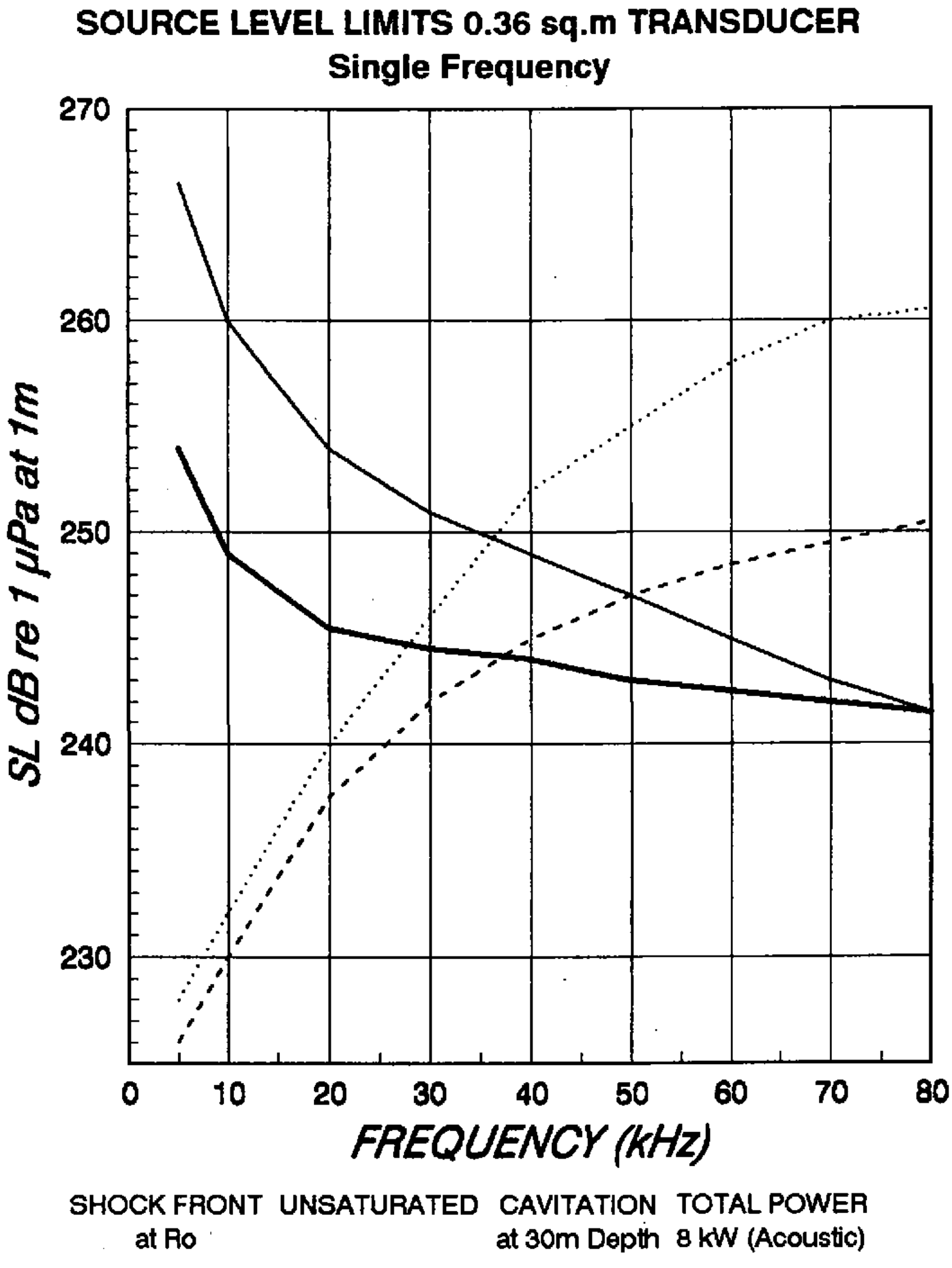

SOURCE LEVEL LIMITS 0.36 sq.m TRANSDUCER
Single Frequency
270
260
250
240
230
SL dB re 1 μPa at 1m
0
10
20
30
40
50
60
70
80
FREQUENCY (kHz)
SHOCK FRONT at Ro
UNSATURATED
CAVITATION at 30m Depth
TOTAL POWER 8 kW (Acoustic)

FIGURE 3

SPECTROGRAMS for DSSC & AM MODULATED SIGNALS

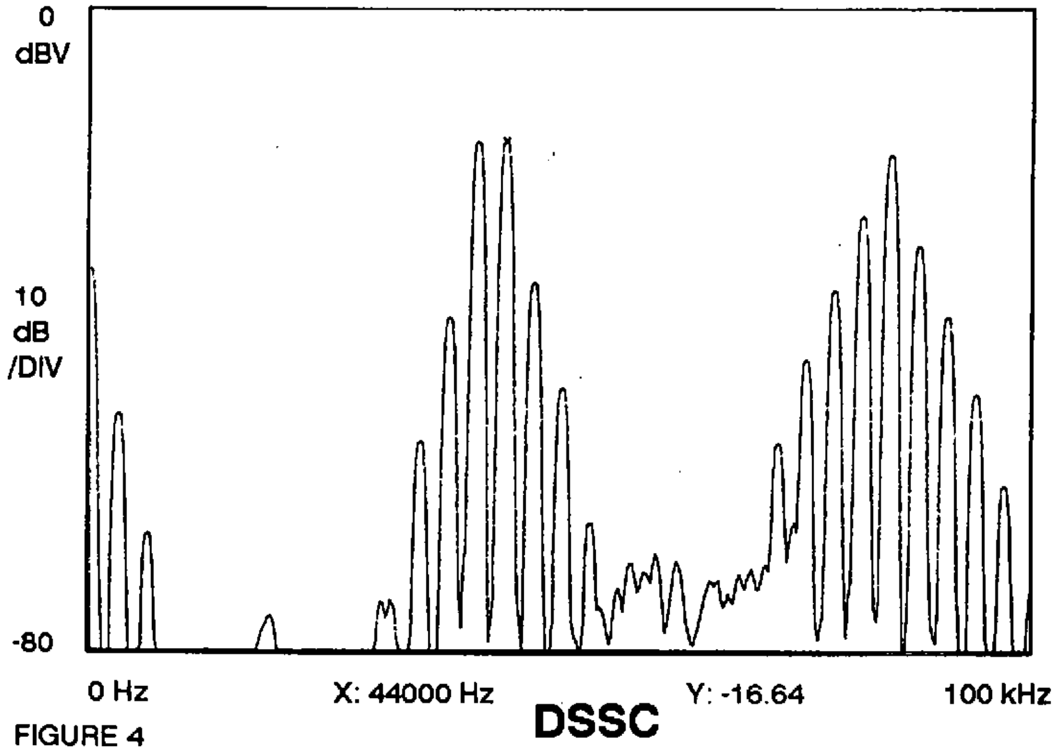


FIGURE 4

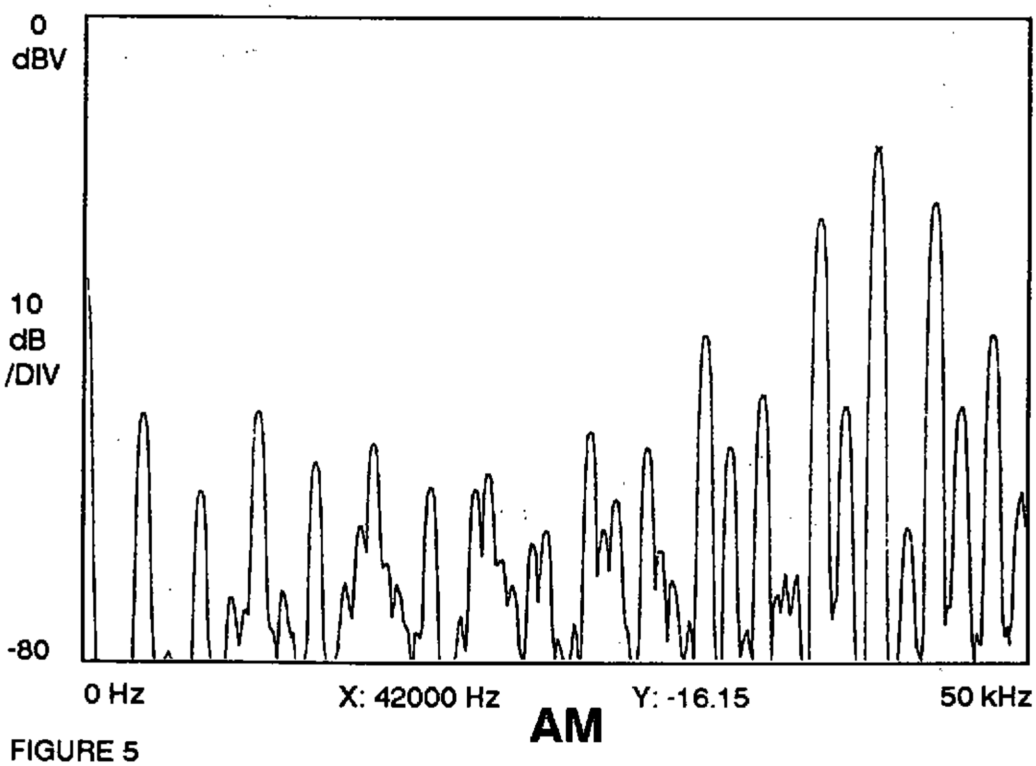


FIGURE 5

EXPERIENCE WITH A VERSATILE NLA SONAR SYSTEM

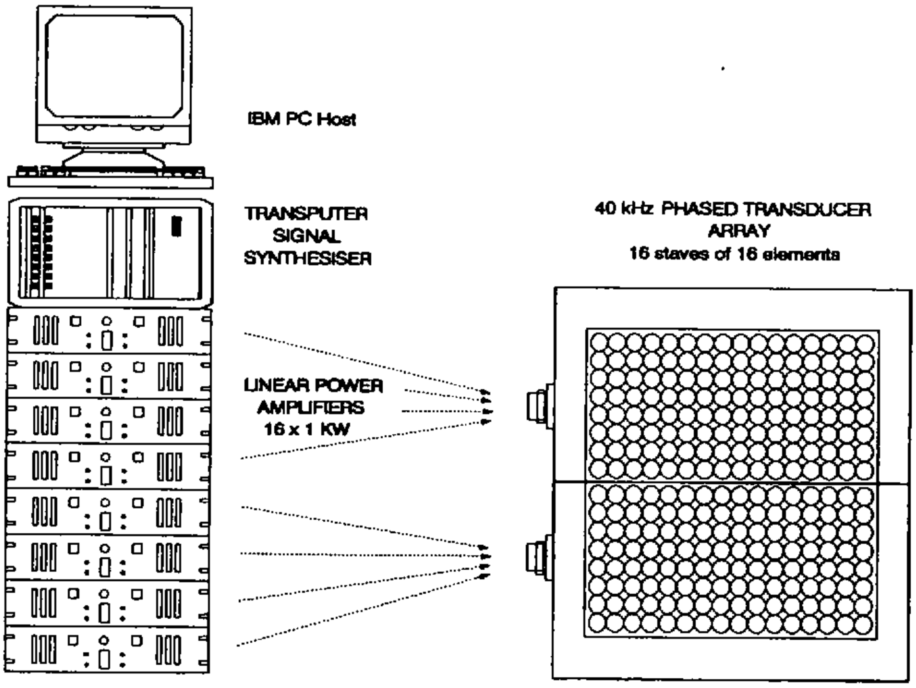


FIGURE 6 TRANSMITTER BLOCK DIAGRAM

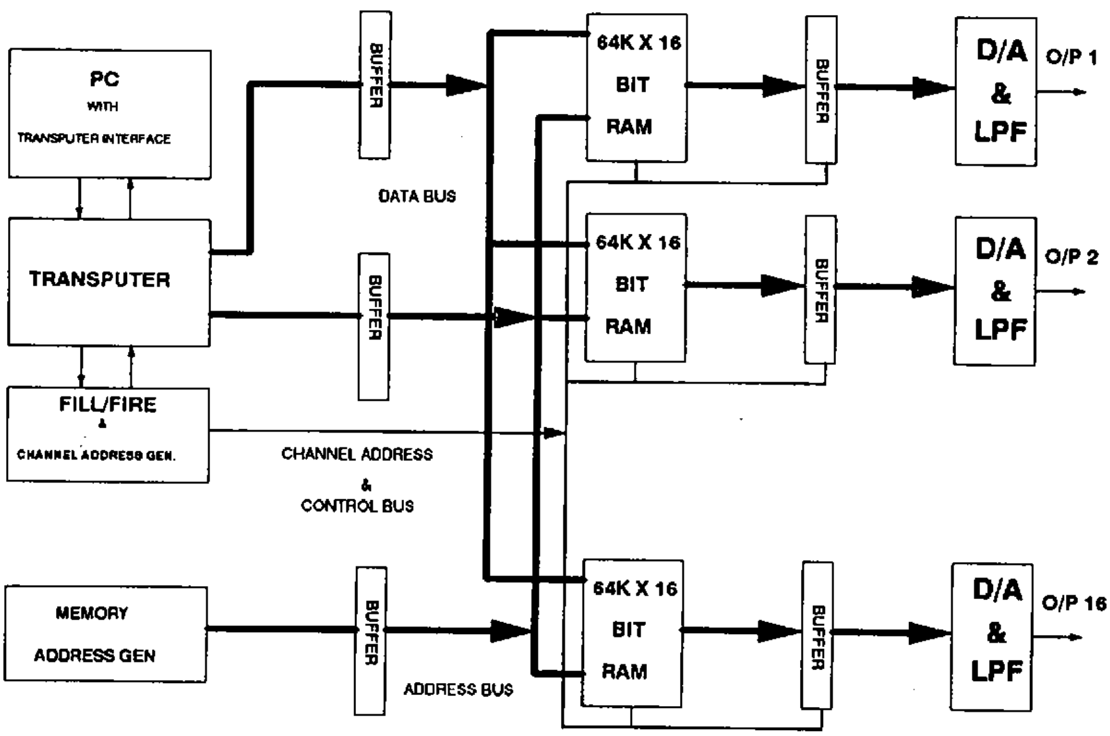
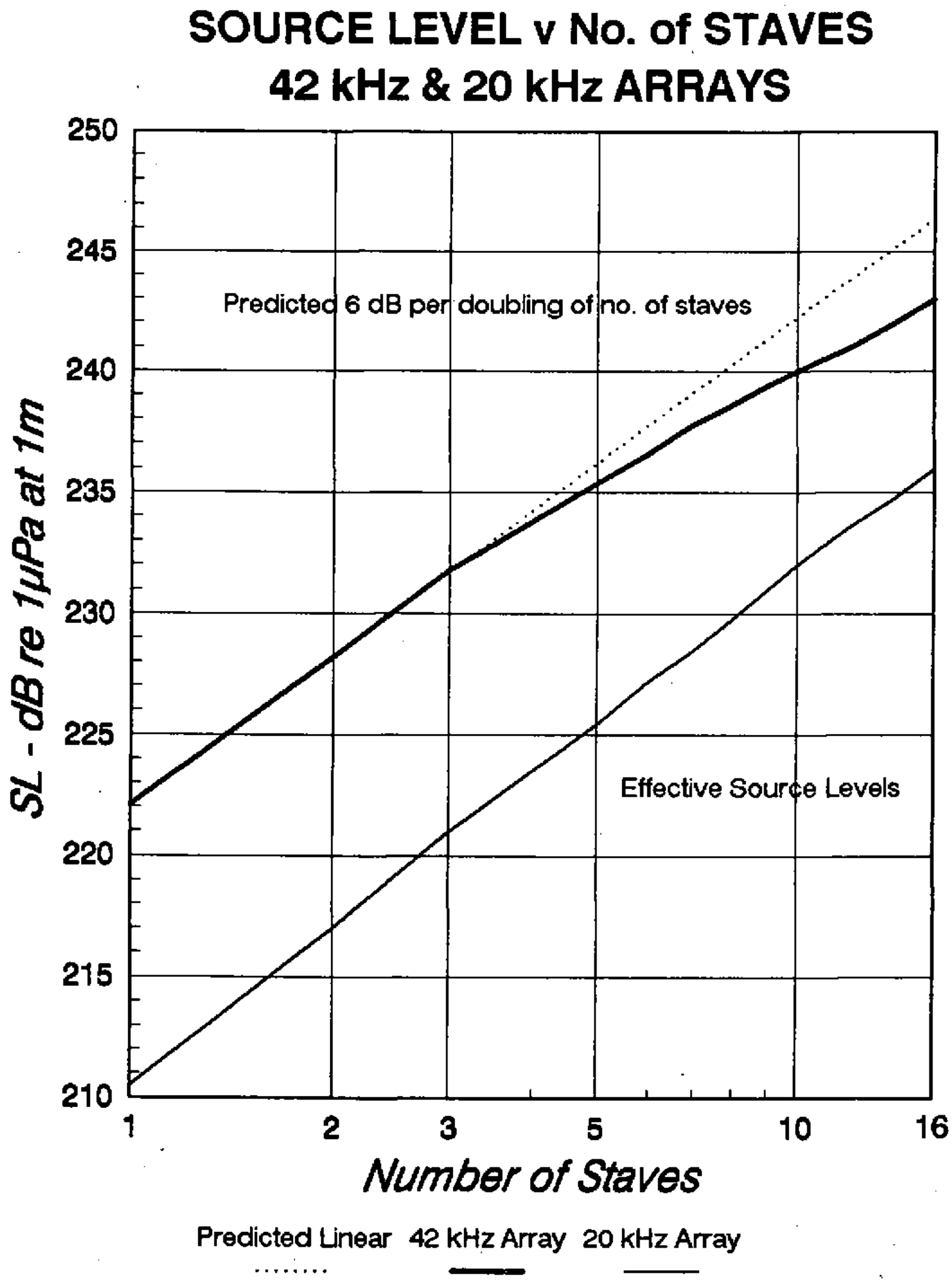


FIGURE 7 TRANSPUTER SIGNAL SYNTHESISER



Hydrophone Range (42kHz) 91.5m
Hydrophone Range (20kHz) 105m
FIGURE 8

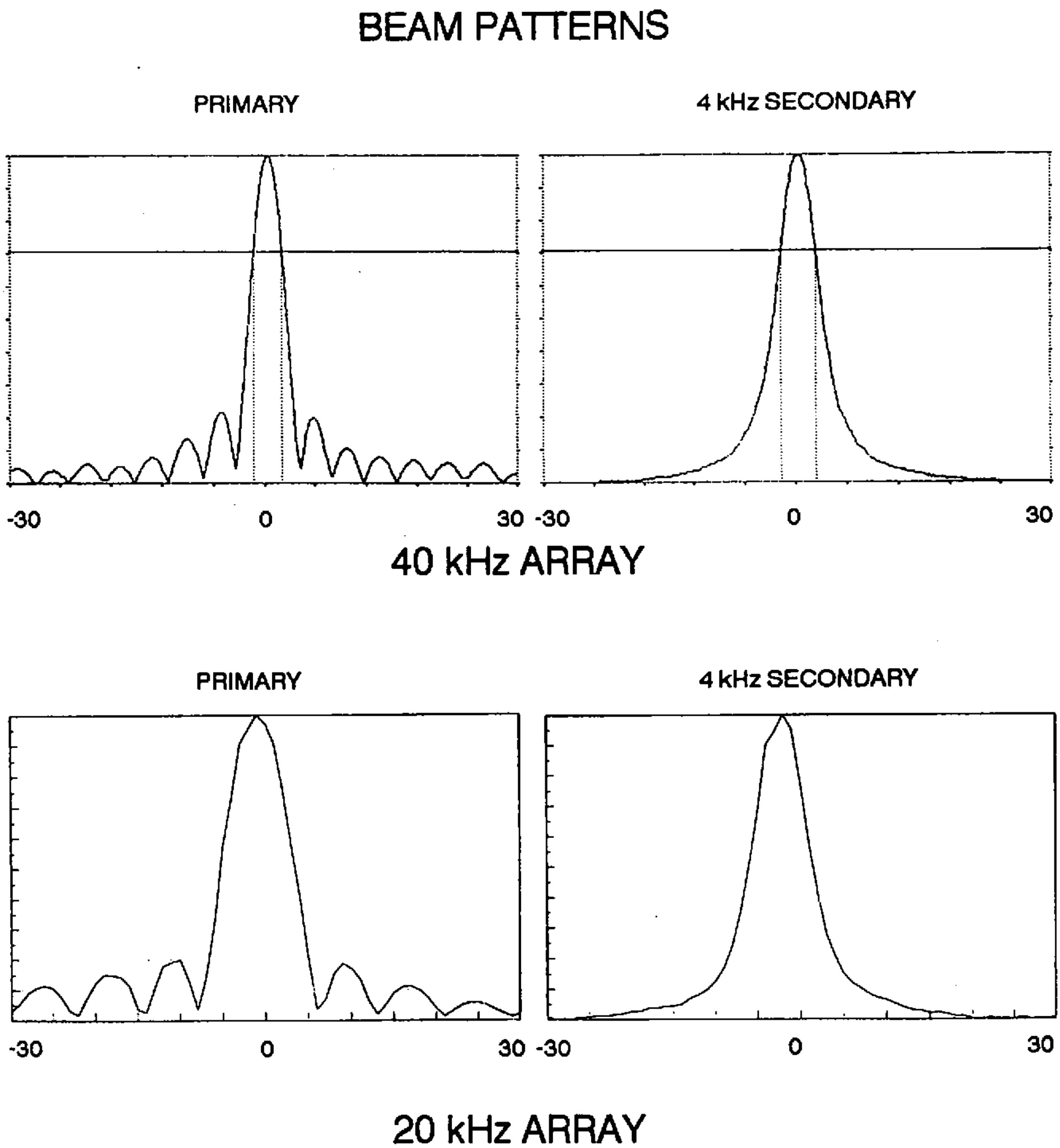

BEAM PATTERNS
PRIMARY
4 kHz SECONDARY
-30
0
30
-30
0
30
40 kHz ARRAY
PRIMARY
4 kHz SECONDARY
-30
0
30
-30
0
30
20 kHz ARRAY

FIGURE 9

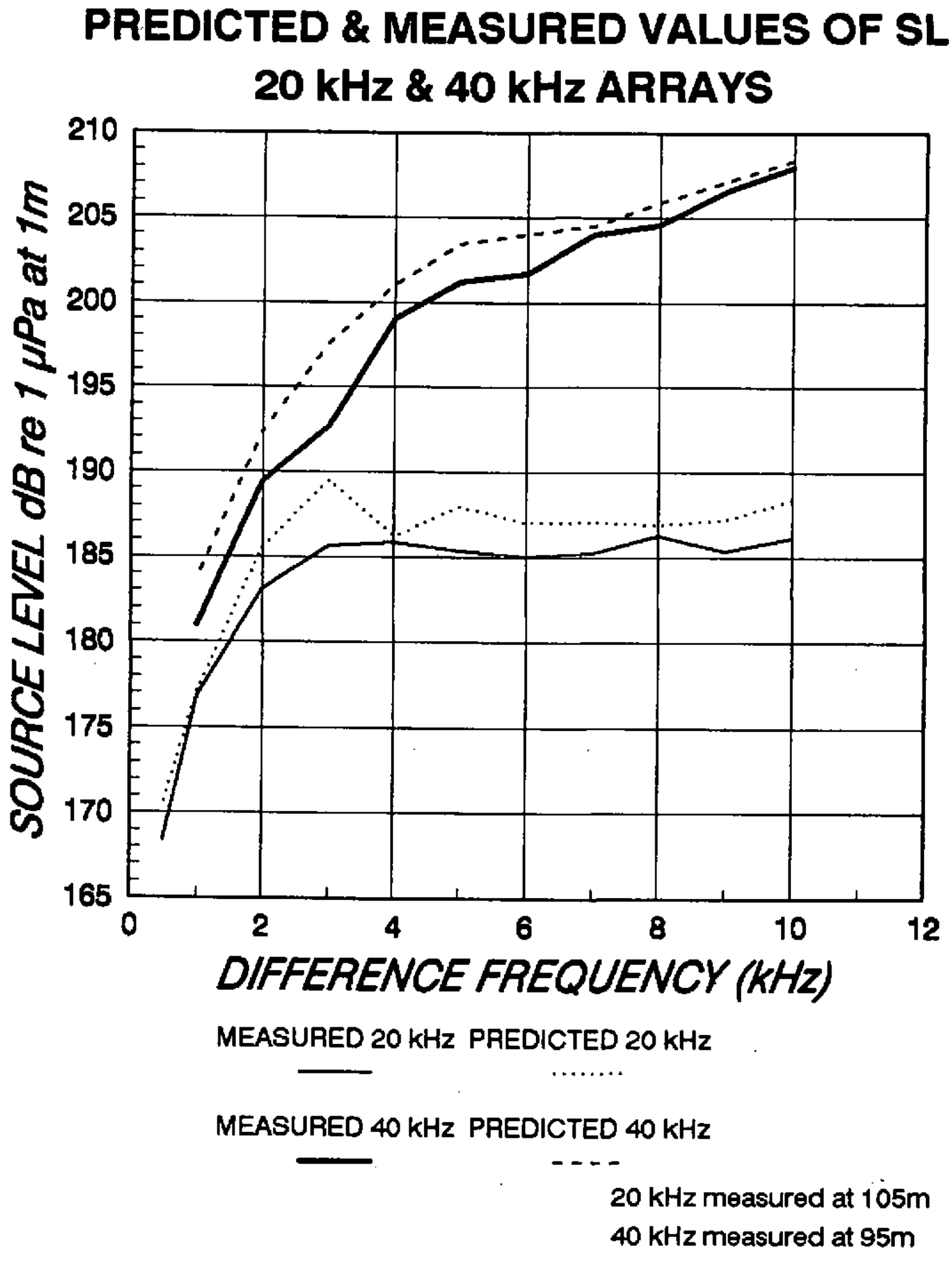
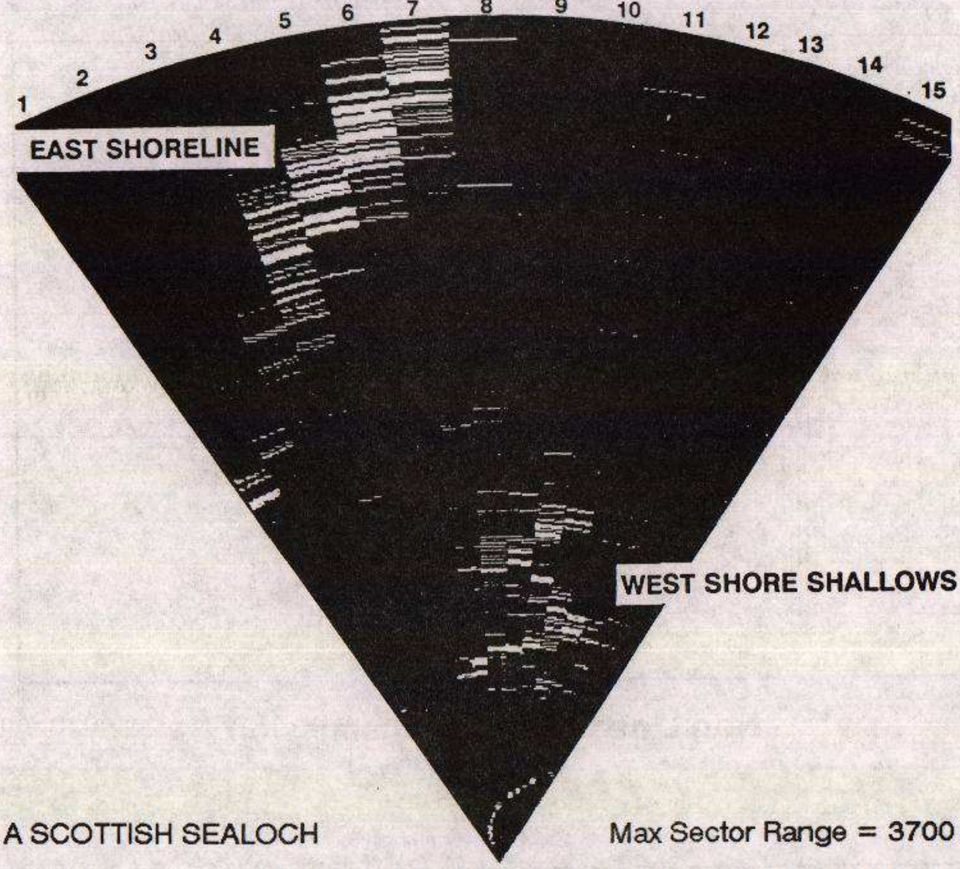

PREDICTED & MEASURED VALUES OF SL
20 kHz & 40 kHz ARRAYS
SOURCE LEVEL dB re 1 μPa at 1m
210
205
200
195
190
185
180
175
170
165
0
2
4
6
8
10
12
DIFFERENCE FREQUENCY (kHz)
MEASURED 20 kHz PREDICTED 20 kHz
MEASURED 40 kHz PREDICTED 40 kHz
20 kHz measured at 105m
40 kHz measured at 95m

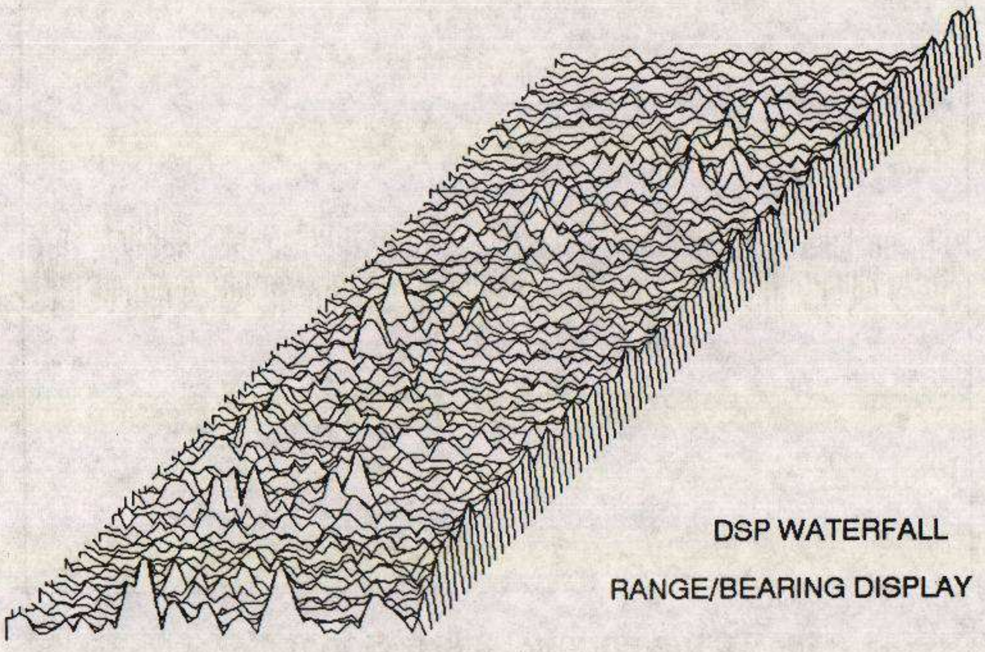
FIGURE 10

EXPERIENCE WITH A VERSATILE NLA SONAR SYSTEM



A SCOTTISH SEALOCH Max Sector Range = 3700 m

FIGURE 11 **40 kHz ARRAY - Tx Coded 2.2 to 3.7 kHz**



DSP WATERFALL

RANGE/BEARING DISPLAY

FIGURE 12 **20 kHz ARRAY - Tx Coded 3 to 5 kHz**

EXPERIENCE WITH A VERSATILE NLA SONAR SYSTEM

DETECTION OF A -6 dB SPHERE

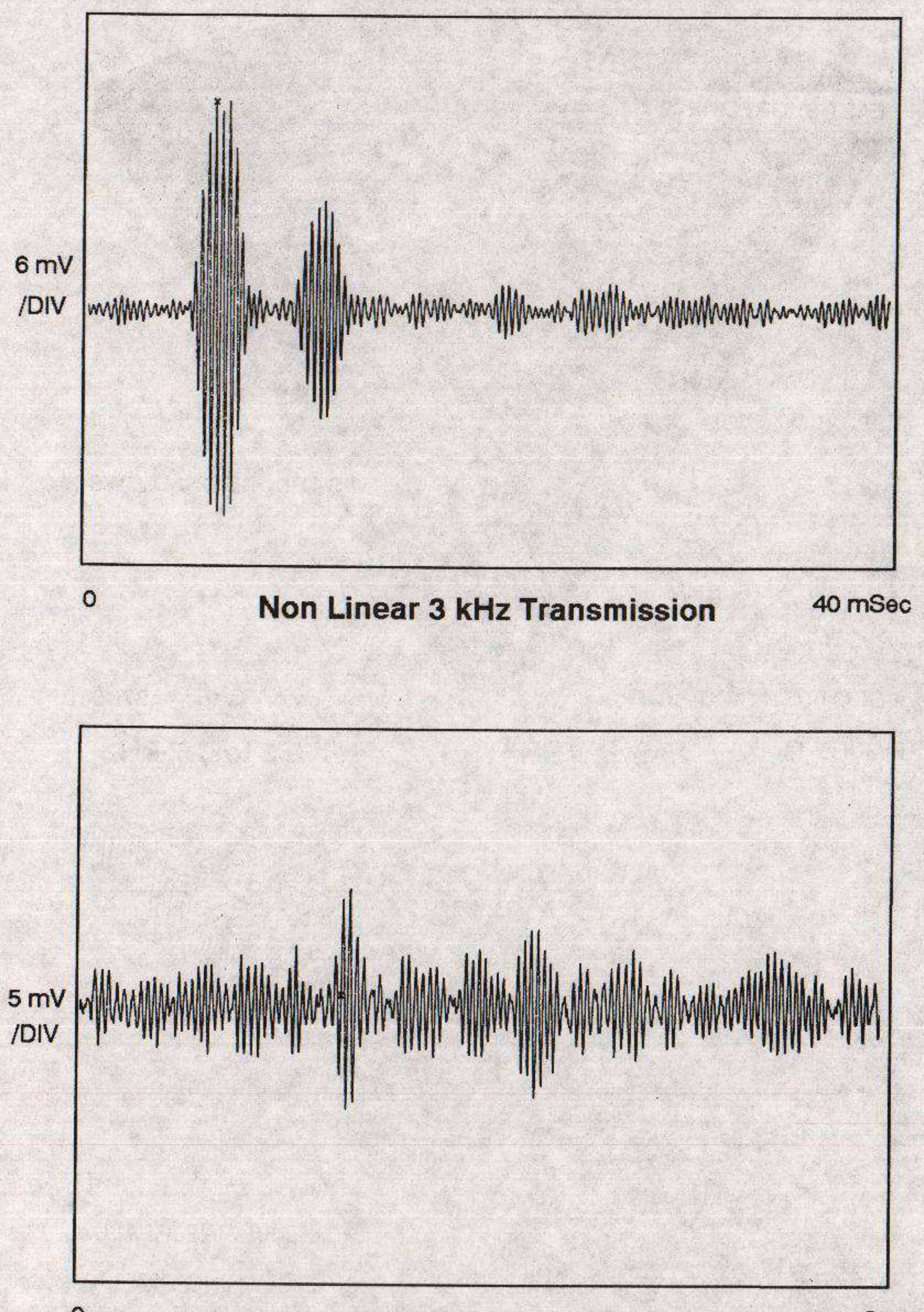


FIGURE 13