WIDE-BAND SOUNDER FOR FISHERIES

M. ZAKHARIA (1), J.P. CORGIATTI (1), F. JOLY (1), R. PERSON (2)

(1) I.C.P.I. Lyon, LTS, URA 346 CNRS, 25 rue du Plat, 69288, Lyon cedex 02, FRANCE
(2) FREMER, Centre de Brest, D.I.T., B.P. 70, 29263, Plouzané, FRANCE

1. INTRODUCTION

Most of the commercial echo-sounders are narrow band systems (single or multiple frequency). It is clear that the use of wide-band systems can help in solving many problems still encountered in fisheries acoustics : size estimation, species identification, detection in reverberating conditions (DSL). In previous papers, we showed experimentally the interest of using such a wide band system to reduce the influence of fish tilt on the size estimation [1] and the possibility of increasing the performance by spectral modelling of echoes [2]. These works were carried out using a relatively wide band system (50 kHz to 75 kHz) developed, at that time, for riverine applications [3], [4].
As no commercial wide band systems were available for developing more sophisticated experiments (and mainly sea experiments) we, I.C.P.I., developped, in collaboration with IFREMER our own system. This system was meant to be both as flexible and as simple as possible for inexperienced users. Sophisticated design and processing is completely transparent for user.
In this paper we present this wide-band sounder prototype (20 to 80 kHz). We will point out the problems encountered while designing such a system and the technical solutions we propose on both acoustical and signal processing aspects.

2. WIDE BAND SYSTEM : ADVANTAGES AND PROBLEMS

Many advantages of using wide-band system have already been pointed out :
- reduction of the influence of fish tilt angle and behaviour on target strength estimation [1]
- possibility of spectral classification [2]
- helping in fighting against reverberation
- possibility of using sophisticated signal processing techniques for target recognition [5], [6], [7] and for fighting against noise (coherent processing)

However a lot of problems will arise while using a wide frequency range :
- transducer design: efficiency and level variations
- transducer beamwidth variations
- absorbtion variations
- increase of the noise level
- real-time processing constraints
- data interpretation.

In the design of our prototype, we tried to find technical solutions (and compromises) to all these problems.

3. GENERAL DESIGN

A schematic view of our sounder is given in figure 1. Three main parts can be pointed out :
-transducer and associated electronics

- power amplifier
- surface unit : pre-processing, signal processing and display.

The transducer and its associated electronics (in waterproof containers) can be installed in a towed fish (better stability, noise avoidance).

Towed body is linked to the power amplifier unit and the surface unit by multi-conductor cables associated to waterproof connectors. The maximum cable length is 30 metres. We will now detail the different elements.

4. TRANSDUCER

The transducer has been designed by Universal Sonar Ltd. It is composed of a square matrix of 100 (10 x 10) identical elements (capable of handling 250 W each). They are associated (cross-connections) in five concentric rings. Each ring is associated to a power amplifier. This allows a beamwidth control over the whole bandwidth. At a first step, a rough control has been done by rings switching. Developments and simulations are presently running to use a frequency variable weighting to smooth the pattern variations.

At this step (and for other processing reasons) the bandwidth has been split in three sub-bandwidth :

- low frequency range: 20 kHz to 40 kHz using 5 rings
- medium frequency range: 40 kHz to 60 kHz using 3 rings
- high frequency range: 60 kHz to 80 kHZ using 2 rings.

The main beam variations (at - 3 dB), without any frequency dependent weighting, will be :

- low frequency range: 10° at 20 kHz, 5° at 40 kHz
- medium frequency range: 9° at 40 kHz, 6° at 60 kHz
- high frequency range: 9° at 60 kHz, 7° at 80 kHz

The waterproof container includes the array connections (100 elements to 5 rings, this connection can be interchanged by simple matrix connector changing), transmitting-receiving diode switchers, low noise pre-amplifiers of echoes and buffers for transmitting echoes through 30 metres of cable (shielded pairs).

The system must be considered as fully wide-band and not as a multiple frequency device. No impedance matching (resonnant system) can be used in this case.

5. POWER AMPLIFIER

The power amplifier is composed of a high voltage power supply and of five identical linear power MOS-FET amplifiers (1 kW each).

These amplifiers can deliver +/- 90 V over a frequency extending from 10 kHz to 100 kHz with an output impedance lower than 0.1 Ω. The maximum transmitted level will be:

- 214 dB re 1 μPa at 1m. at 30 kHz
- 215 dB re 1 μPa at 1m. at 50 kHz
- 207 dB re 1 μPa at 1m. at 70 kHz

As long as we are using linear power amplifier, the level variations over the whole frequency range can be corrected via inverse filtering and pre-accentuation.

We showed this possibility in a previous work in higher frequency range [8] ; the application to this prototype is in project ; corrected transmitted signals will be computed and stored in EPROM memory. It is important to notice that frequency pre-filtering and beamwidth control are made possible by the use of a wide-band linear power amplifier.

6. PROCESSING UNIT

The processing unit is described in figure 2. Transmitted signals are stored in digital memory (EPROM). For each frequency range, three signal duration are possible : 1ms, 5 ms and 10 ms. The signal selection (bandwidth and duration) is made from front panel.
The front panel control will automatically set up transmitting multiplexer as well as receiving multiplexer in the right configuration.
Signal processing is done on two levels :
- analog processing separate for each frequency range
- digital processing common to all frequency ranges.

Analog processing consists in :
- band-pass filtering
- time varying gain: $10 \lg r + \alpha.r$, $20 \lg r + \alpha.r$, $30 \lg r + \alpha.r$ or $40 \lg r + \alpha.r$
- frequency shifting: 20 kHz, 40 kHz or 60 kHz

The attenuation coefficient α is chosen according to the carrier frequency (for each frequency range). At 100 m, the maximum compensation error is less than 0.5 dB (for all ranges).
All the correction laws are digitally computed and stored in EPROM memory.
The signal design (frequency bandwidth and carrier frequencies) is such that the matched filter input signal is independant of frequency range. Then, only 3 matched filters are needed (one for each signal duration, frequency range: 0.5 kHz to 20 kHz).
With such a design and such a processing, time resolution at the output of the matched filter will be the same for all the transmitted signals : any frequency range, any duration.
The axial resolution (at -6 dB) is about 7.5 cm, for all ranges; this will be very helpful for data interpretation and display. All the processing is transparent for the user who can choose signal duration, frequency range, TVG law, repetition rate and observation window on front panel. This panel is controlled by a microcontroller µC8031 which takes care of all the signal timing and multiplexing.

7. REAL TIME DIGITAL FILTER

Matched filter is achieved digitally (F.I.R.). The digital filter is based on AT & T DSP 16 signal processor.
The processing board (double Europe board) contains:
- 2 A/D converters: 12 bits 50 kHz
- 2 D/A converters: 12 bits 50 kHz
- 2 DSP 16 with associated memory (cycle time: 55 ns)
- 1 controller with associated memory (NEC V25).

The controller gives an external access to the board via RS232 serial link. It is mainly used to load the filters responses and to store them in its memory (EPROM or EEPROM) for stand alone operations.
In a stand alone configuration, the controller selects the right filter response, loads it in the master signal processor memory and sets up the converters configuration. It is then disconnected from DSP processors which will run in real time.

For our application 2 DSP 16 are used (1 master and 1 slave). The configuration can be extended, on board, to 4 DSP 16 (1 master, 3 slaves) . It can also be extended (if needed) to 8 DSP 16, with add-on board. With two DSP chips, the real-time processing performance limits the signal duration to 10 ms for a sampling frequency of 40 kHz.

The transmitted signals expression is :

$$S(t)= A.\exp[-\ln^2(t/t_0) / \ln g] . \cos[2\pi b\ln(t/t_0) / \ln g]$$

It has mainly been chosen in accordance to works we are developing, in parallel, on target recognition and classification [5] [6] [7]. Figure 4 is an illustration of such a signal. Figures 5 shows both a transmitted signal (top) and a real-time processing output (bottom) for various durations and frequency ranges. These results (obtained in laboratory conditions) show that the output characteristics -processing delay, time resolution- do not depend on the operating conditions. This has been achieved via signal design to make the processing transparent for users.

8. CONCLUSION

We showed, in the prototype description, that most of the sonar problems encountered while using wide-band systems can be solved via digital signal processing techniques. Some of them are already implemented (coherent processing), others (transducers bandwidth compensation and beam control) are under development and are made possible by computing transmitted signals and using linear power amplifiers. Further calibrations and systematic experiments on fish (various species and sizes) together with echoes processing and parameters extraction (size estimation and species recognition) are now planned in controlled situation (video control and image processing).

9.REFERENCES

[1] M.ZAKHARIA: "Variations of fish target strength induced by its movement; a wide-band impulse experiment.", Int. Symposium on Fisheries Acoustics, Seattle (1987). (to be published in : Rapp. P.-V. Réun. Cons. Int. Expor. Mer).

[2] P. DEGOUL, P. FLANDRIN, N. GACHE, M. ZAKHARIA : "Fish Echoes Classification via Auto-Regressive Modelling." Proc. IOA «Progress in Fisheries Acoustics», vol. 11, n° 3, pp. 161-168, Lowestoft, U.K. (1989).

[3] M.ZAKHARIA : "A Prototype of Wide-Band Sonar for Fisheries in Lakes and Rivers". Int. Symposium on Fisheries Acoustics,(1987). (to be published in : Rapp. P.-V. Réun. Cons. Int. Expor. Mer).

[4] M.ZAKHARIA, J.M.PEY : "Détection et comptage de poissons par sonar haute fréquence.Possibilités de reconnaissance des espèces et d'estimation de la taille." 9° colloque GRETSI-83, pp. 753-761, Nice 1983.

[5] M.ZAKHARIA : "Contribution à la caractérisation et à l'identification de formes simples par sonar actif : application à un sonar de pêche." Thèse de Docteur-Ingénieur, spécialité: acoustique, Faculté des sciences de Luminy, Université d'Aix-Marseille II.1982

[6] M.ZAKHARIA, J.P. SESSAREGO : "Sonar target classification using a coherent echo processing."IEEE, ICASSP-82, pp.331-334, Paris.1982

[7] R.A. ALTES : "Sonar for a generalized target description and its similarity to animal echolocation systems". J.Acoust.Soc.Am. 59, 97-105 1976.

[8] M.ZAKHARIA : "Ultrasonic Transducer Response Correction via Inverse Filtering. Characterization in the Time-Frequency Plane". Ultrasonics International, Madrid July 1989.

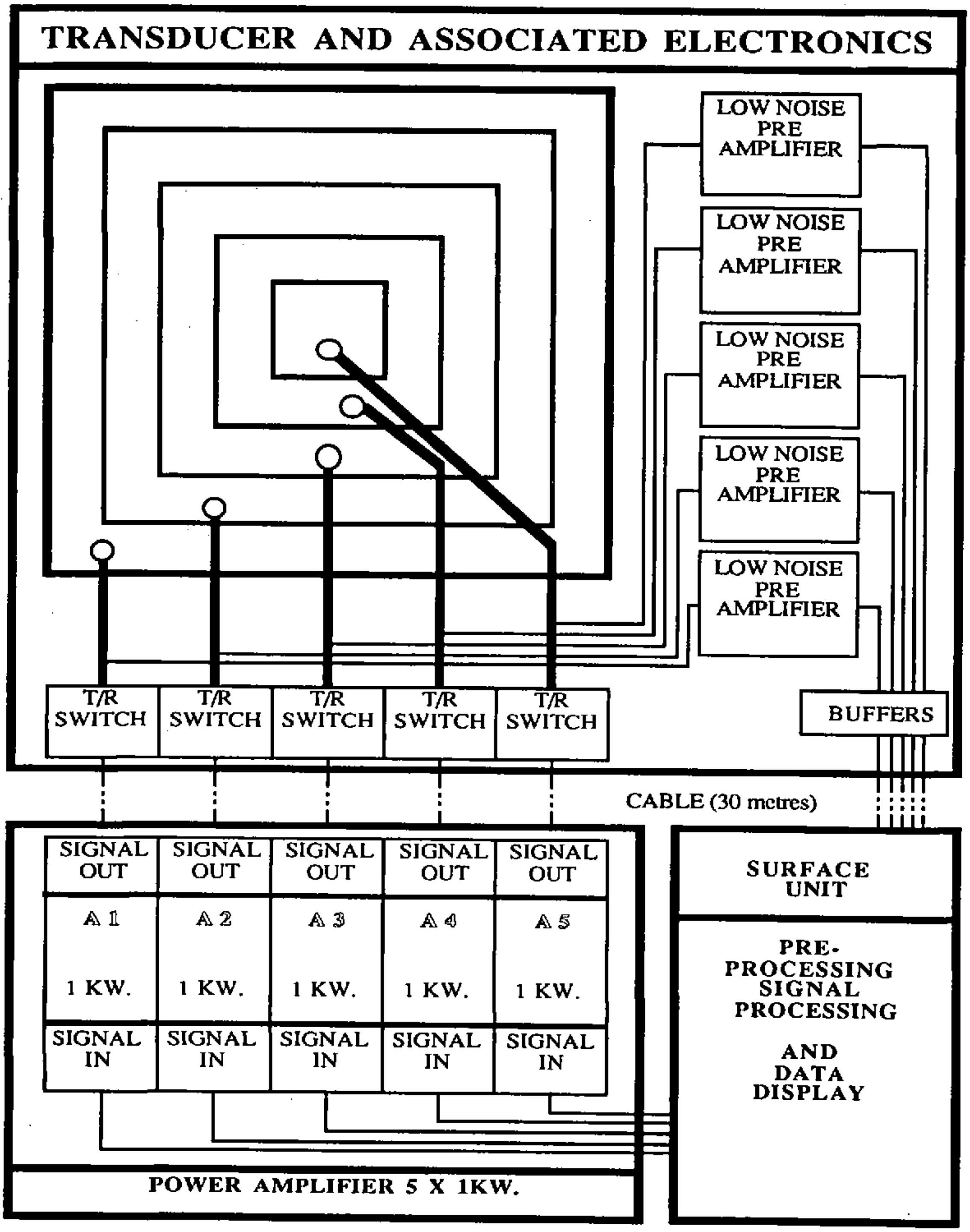


FIGURE 1 : TRANSDUCER AND ASSOCIATED ELECTRONICS

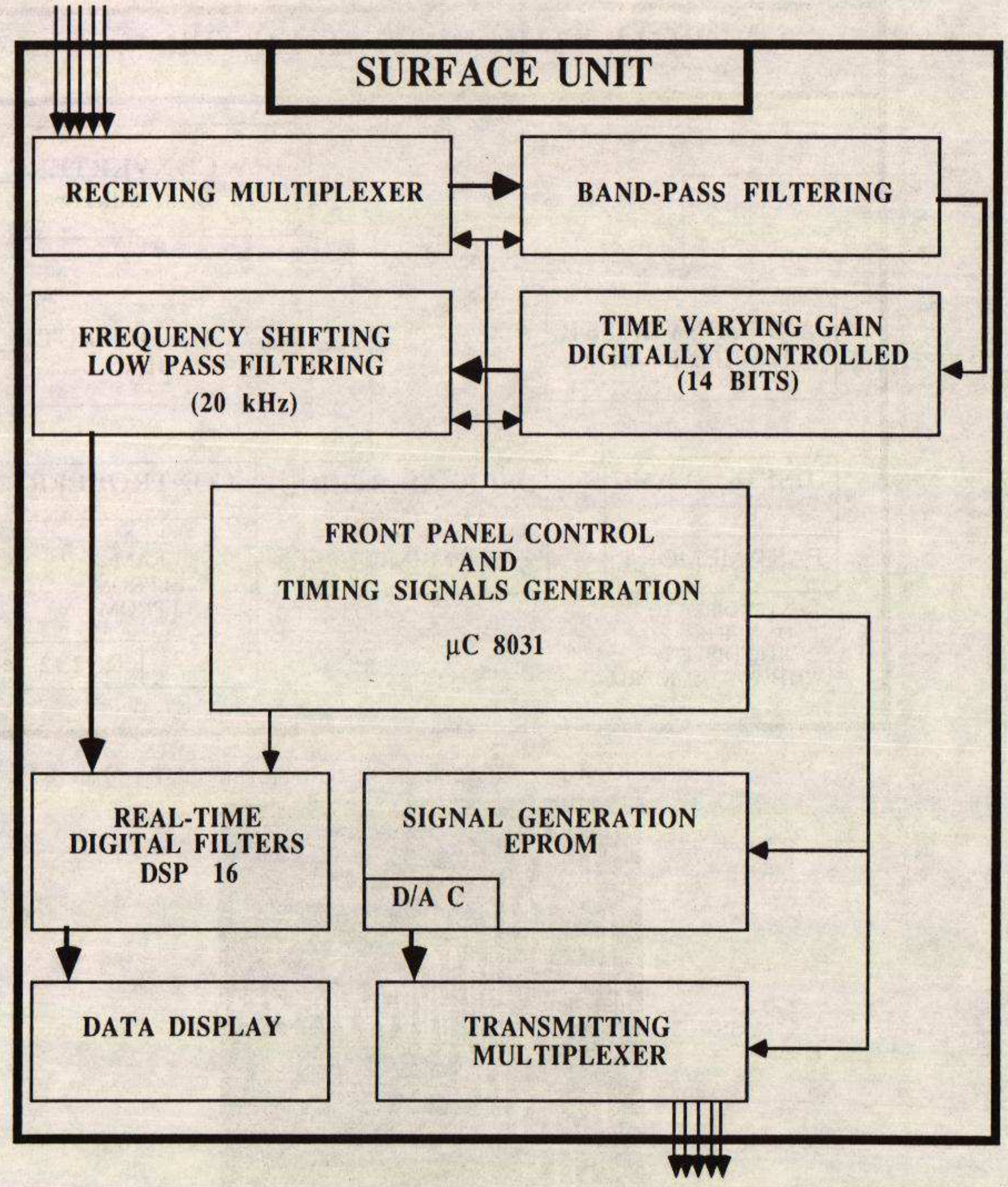


FIGURE 2 : SURFACE UNIT

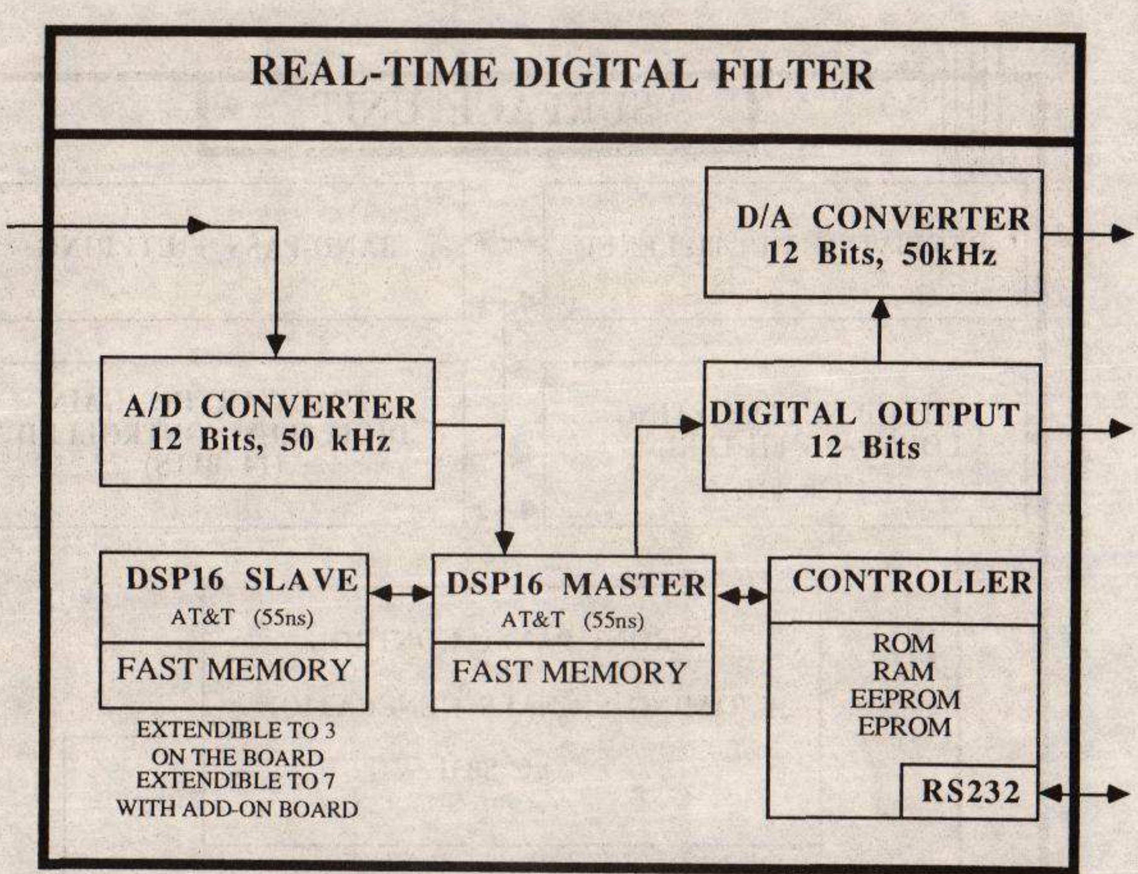


FIGURE 3 : REAL-TIME DIGITAL FILTER

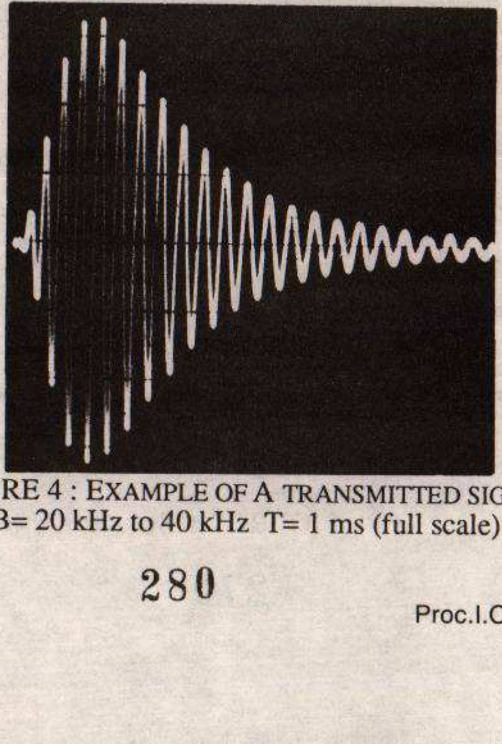
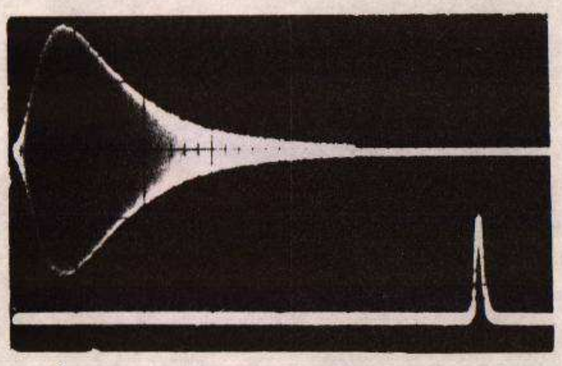
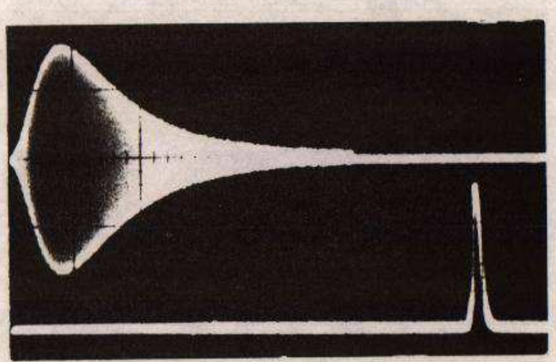


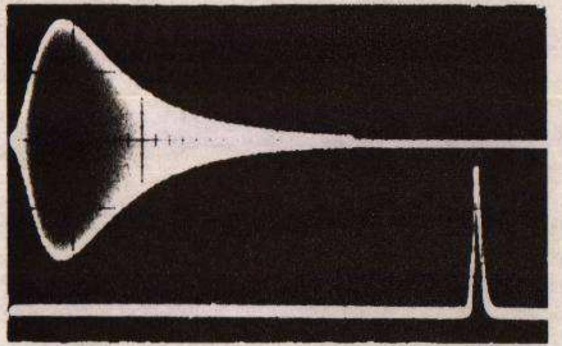
FIGURE 4 : EXAMPLE OF A TRANSMITTED SIGNAL
B= 20 kHz to 40 kHz T= 1 ms (full scale)



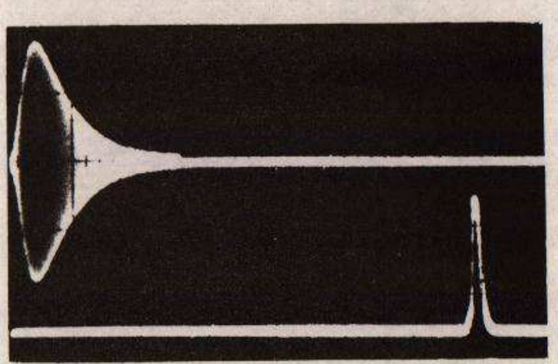
-a-



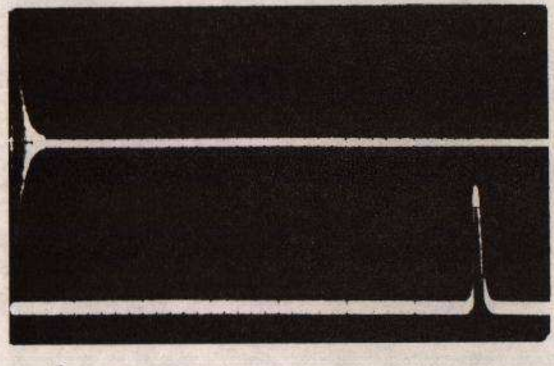
-b-



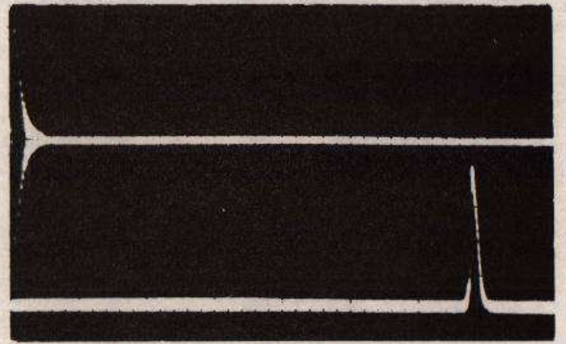
-c-



-d-



-e-



-f-

FIGURE 5 : PROCESSING EXAMPLES
(top: transmitted signal, bottom processing output, same scale: 2ms/div, T: signal duration)
output is unchanged for all operating conditions : frequency range, signal duration

a: Low Frequency: 20-40 kHz, T=10ms
c: Medium Frequency: 40-60 kHz, T=10ms
e: High Frequency: 60-80 kHz, T= 1ms

b: High Frequency: 60-80kHz, T=10ms
d: Medium Frequency: 40-60kHz, T= 5ms
f: Low Frequency: 20-40kHz, T= 1ms