Modal Processing for Acoustic Communications in Shallow Water Experiment

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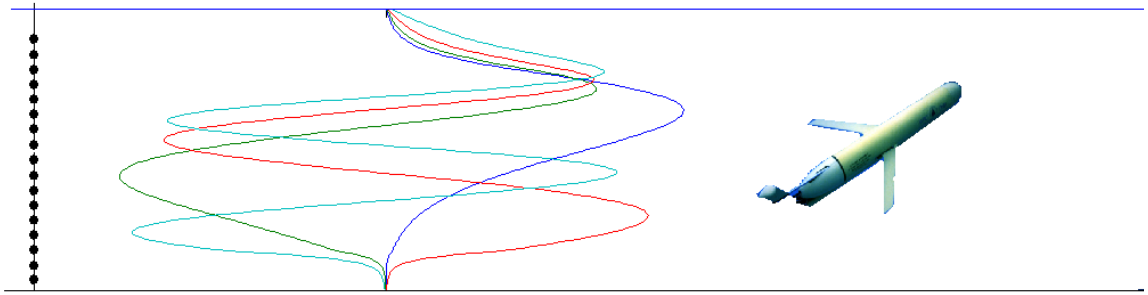
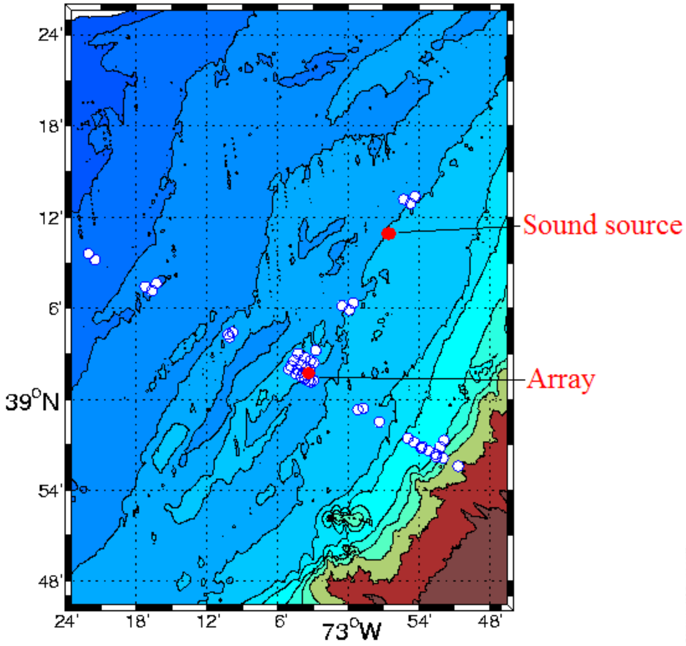
OBJECTIVE

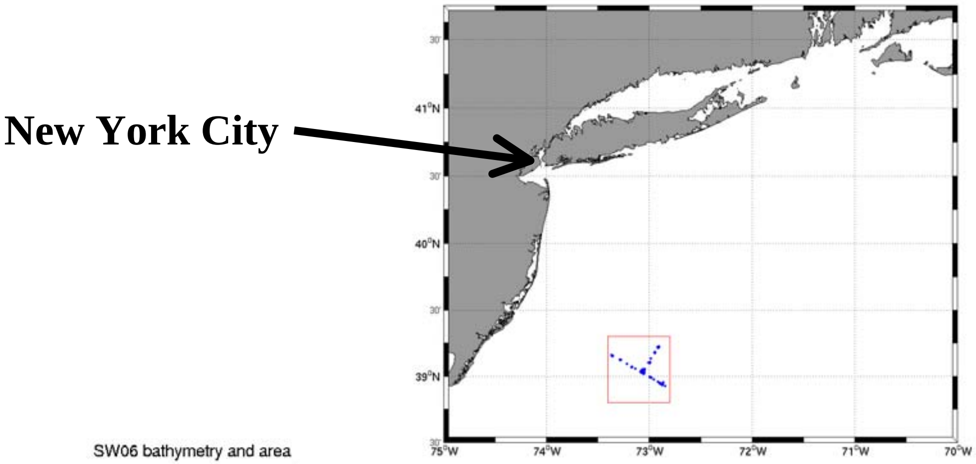
- To process and analyze PSK m-sequence signals with different carrier frequencies transmitted a distance of 19.2 km and collected by an array during the Shallow Water 2006 experiment.
- To show the feasibility of broadband mode decomposition as a preprocessing method to reduce the effective channel delay spread and concentrate received signal energy in a small number of independent channels.
- To show that this method is reliable and stable and even during strong internal wave activity a low bit error rate can be achieved.

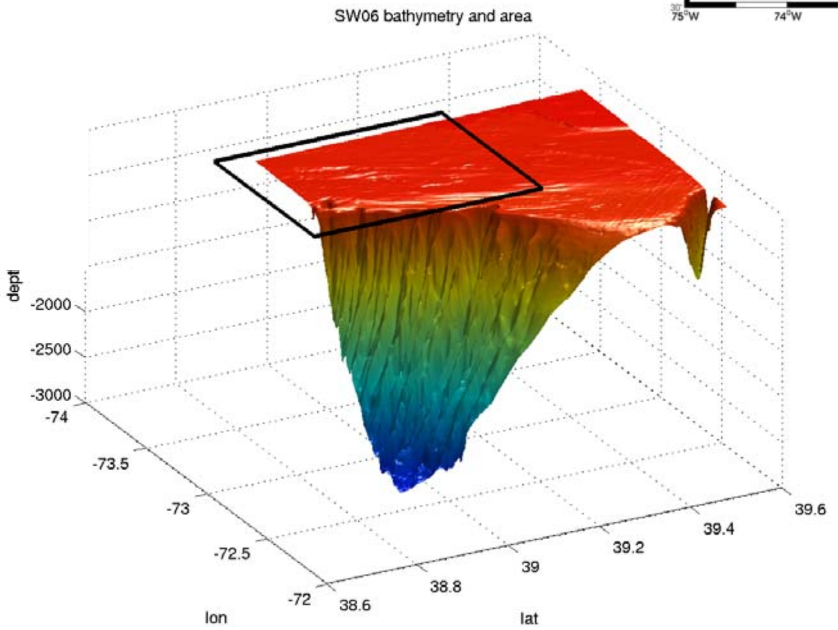


Shallow Water Experiment (SW06)

- **Location, bathymetry, source and receiver positions.**

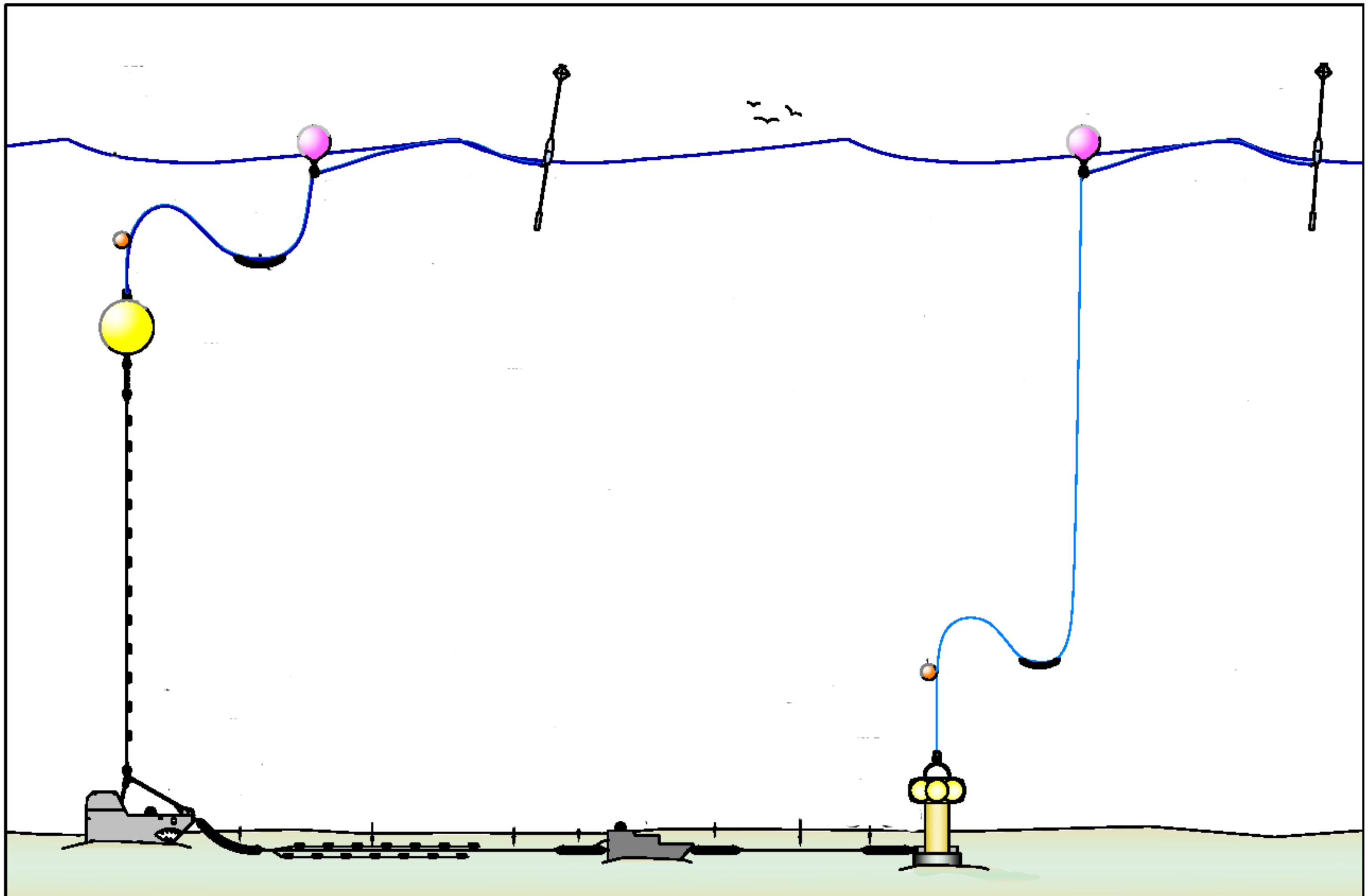






Acoustical Array "Shark".

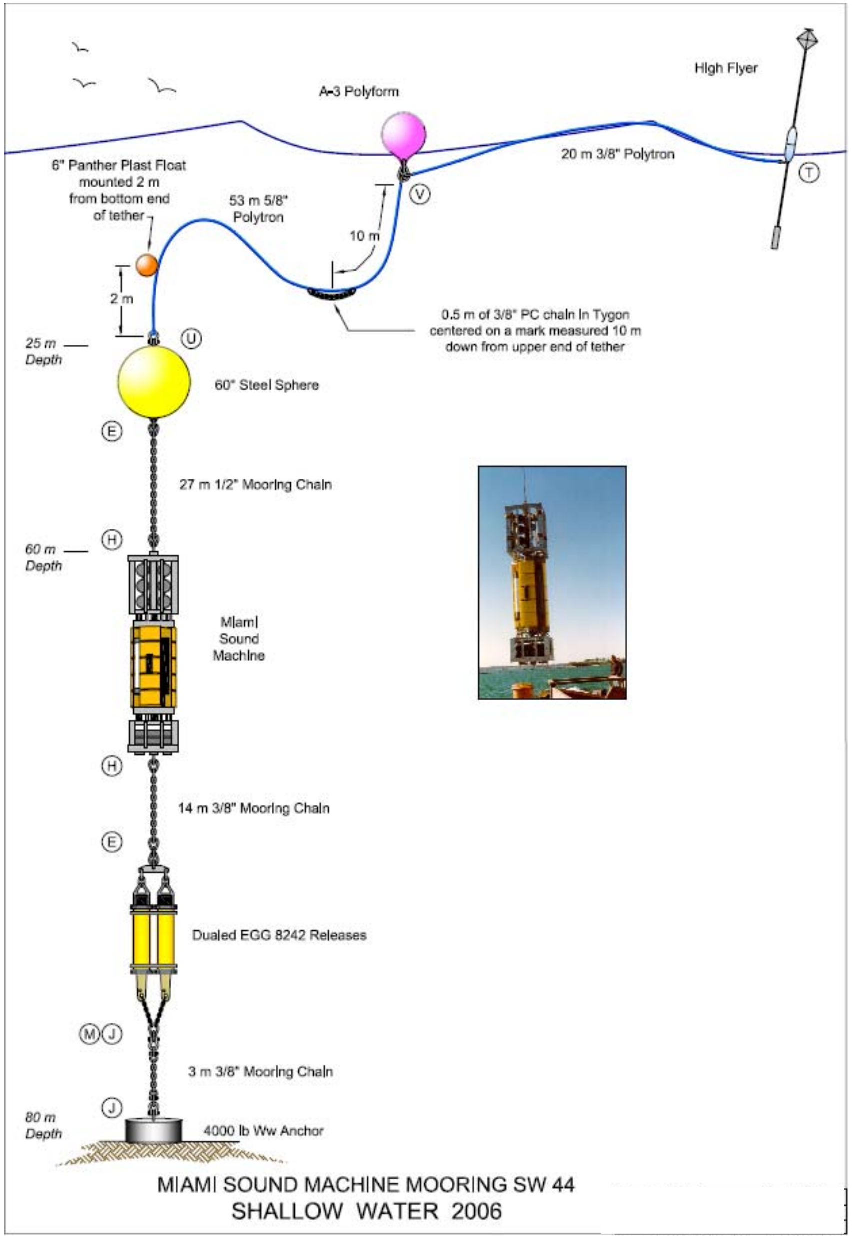
- The array system included a 16 channel VLA and a 32 channel HLA. The system was deployed in 78 m of water, which allowed 14 of the 16 vertical array channels to span the water column from about 76 m to 12 m depth.



SHARK VLA - HLA Mooring SW 54, SHALLOW WATER 2006

Low Frequency Sound Source

63 bit M-sequence 101.7 Hz;
127bit M-sequence 203.4 Hz;
511bit M-sequence 813.8 Hz.
PSKM 180 deg, 4 carrier cycles per digit, 40 M-sequence periods



ACOUSTIC FIELD NORMAL MODE DECOMPOSITION

$$p(r,z)=\frac{i}{p(z_s)\sqrt{8\pi r}}\exp(i\pi/4)\sum_{m=1}^{\infty}\Psi_m(z_s)\Psi(z)\frac{\exp(ik_{rm}r)}{\sqrt{k_{rm}}}$$

Signals from hydrophone array.

$$p(z_k,f)=\sum_{m=1}^{\infty}A_m(f)\Psi_m(z_k,f)+n(z_k,f),\quad P = \Psi A+N$$

- Direct projection $\mathrm{A}=\Psi^T P$
- Moore-Penrose pseudo-inverse transformation $\mathrm{A}=(\Psi^T\Psi)^{-1}\Psi^T P$
- Total Least Squares (TLS) $[\Psi\,|\,P]=U\Sigma V^T, V=\begin{bmatrix} V_{11} & V_{12} \\ V_{21} & v_{22}\end{bmatrix} \Rightarrow A_{TLS}=-V_{12}/v_{22}$
- Minimum Variance Distortionless Response (MVDR)
- Maximum a Posteriori Probability (MAP)

The pseudo-inverse was used to perform the modal filtering for each frequency of signal section spectrum.

The resulting filtered signals were transformed to low frequencies.

After inverse Fourier Transform the processed sections then were down-sampled and combined by overlap-add method.

Another approach based on the low-pass equivalent method (base-band mode filtering) was tested and gave the equivalent result with less computation expenses.

MAXIMUM LIKELIHOOD DETECTOR WITH JOINT CHANNEL ESTIMATION AND DATA RECOVERY

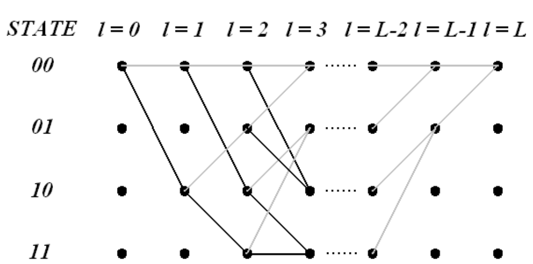
Joint channel response estimation and data recovery (pre-survivor processing).

The comparison analysis has shown the superior performance of the MLSD algorithm with a joint channel estimation and data recovery for each sequence of data symbols then the traditional DFE equalizer in a channel with severe frequency selective fading.

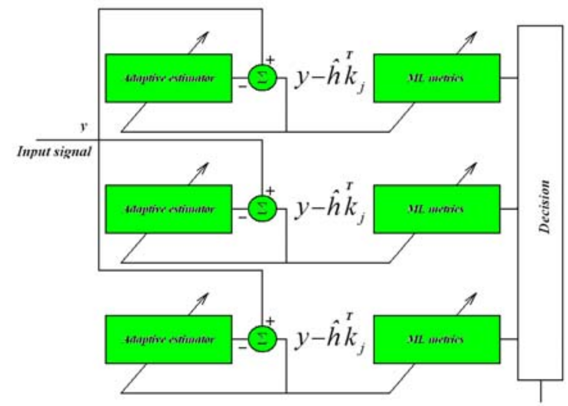
The simplified equations for the maximum-likelihood (ML) metrics w_l are

$$w_l = w_{l-1} - |d_l|^2, \quad \hat{h}_{lm} = \hat{h}_{l-1,m} - a d_l k_{l-m}, \quad d_l = y_l - \sum_{m=0}^{N} \hat{h}_{lm} k_{l-m}$$

$\hat{h}_{lm}$ are minimum LMS estimations of channel impulse response of low frequency equivalent; $k_j = \pm 1$ is the data sequence; y_l is the input data; a = 0.1 is a step size of estimation algorithm.

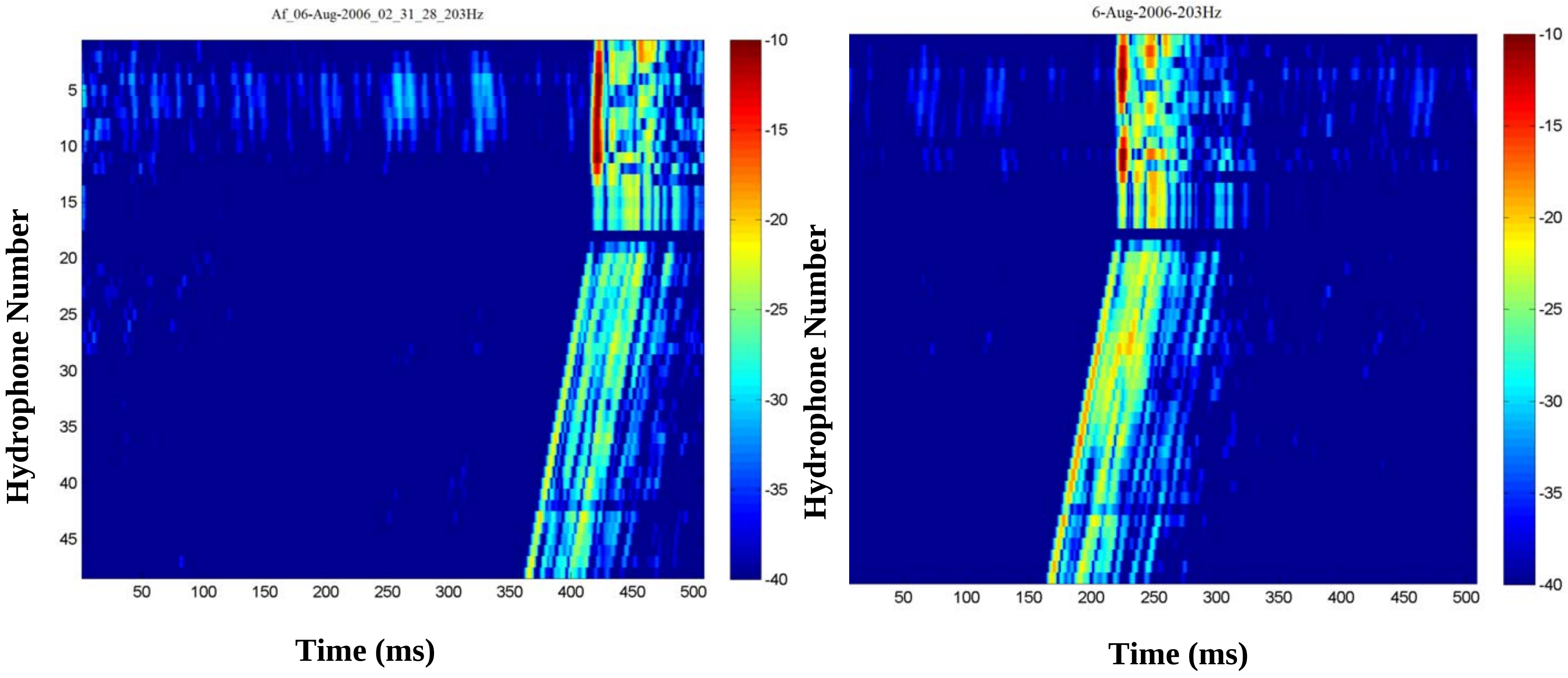


Trellis for a four-state shift-register process



Block-diagram for adaptive algorithm of metrics calculation.

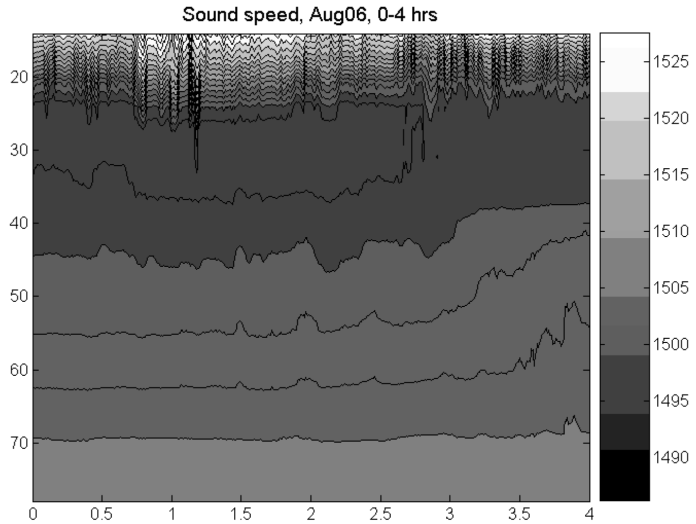
PULSE COMPRESSION OF ARRAY INPUT SIGNALS

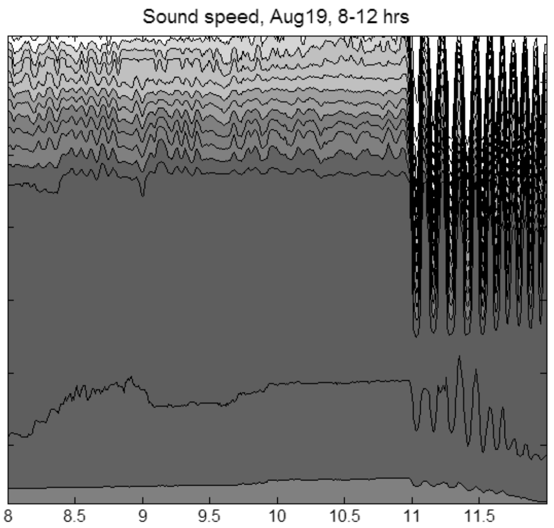


T. F. Duda and J. M. Collis, Acoustic field coherence in four-dimensionally variable shallow water environments: estimation using co-located horizontal and vertical line arrays, Proc. 2nd Meeting Underwater Acoustic Measurement Conf., Heraklion, Greece (2007).

SOUND VELOCITY IN WATER AND BOTTOM

Depth (m)



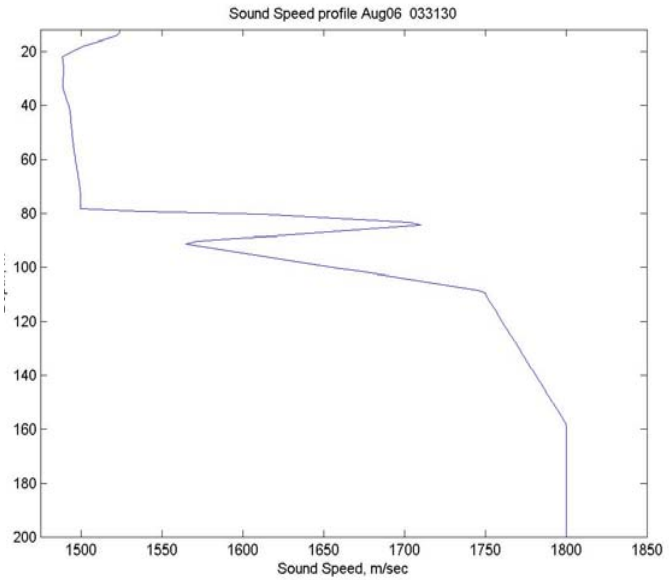


Time (hours)

Time (hours)

Sound speed fluctuations

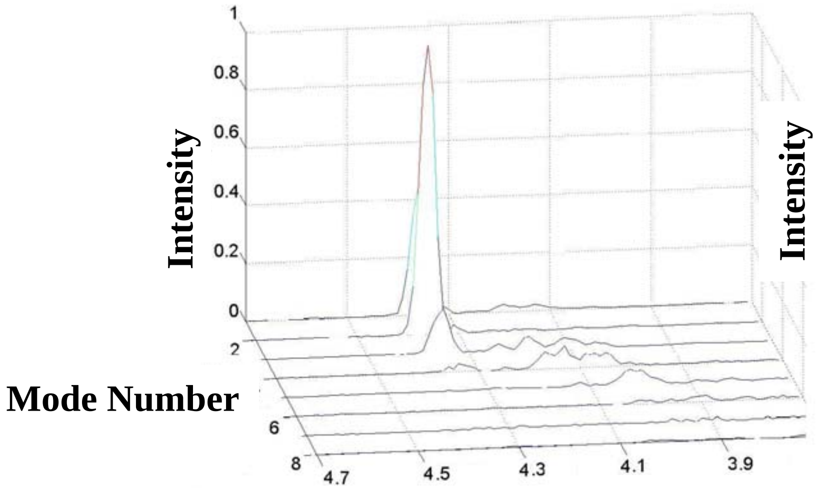
Depth (m)



Sound speed (m/s)

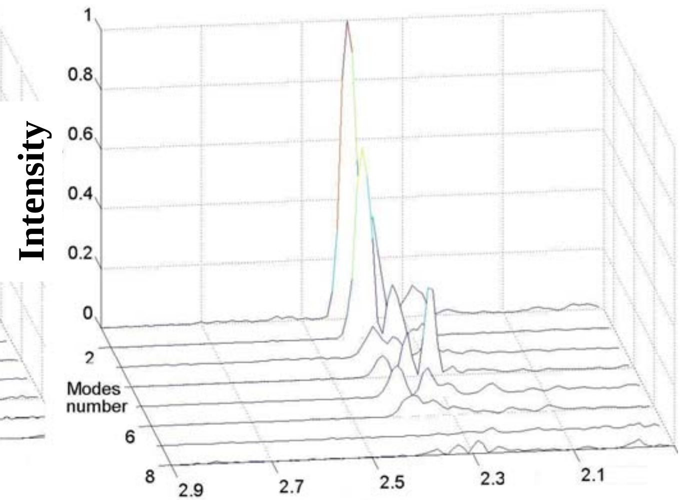
PULSE COMPRESSION AFTER MODE FILTERING

August 6



A **Time (s)**

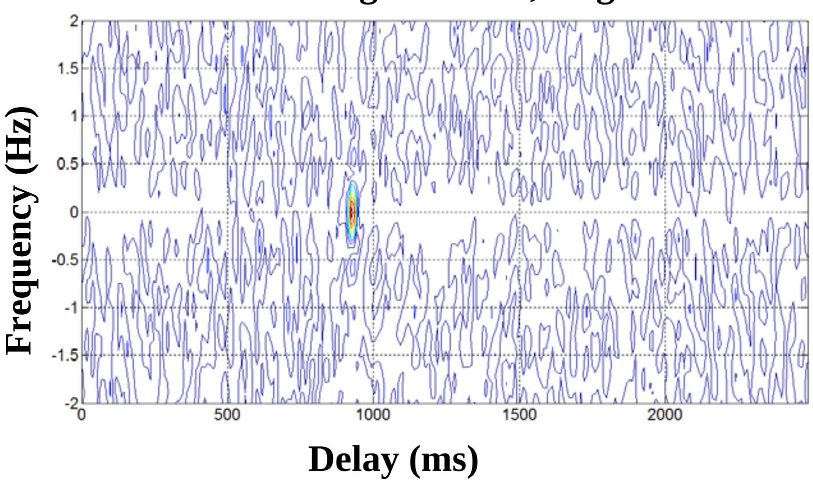
August 19



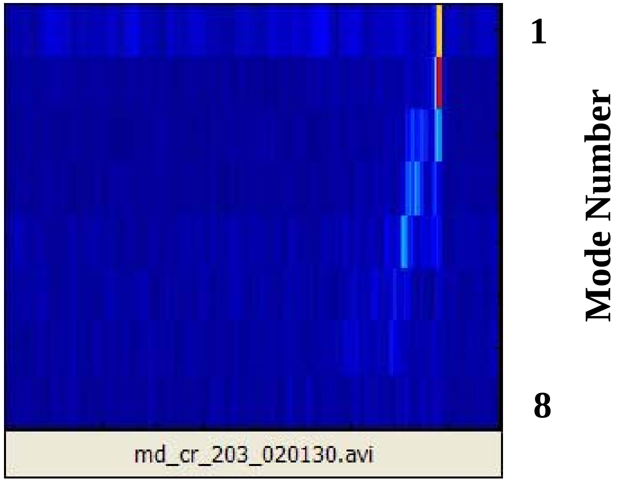
B **Time (s)**

Mode-time correlations

Scattering Function, August 19

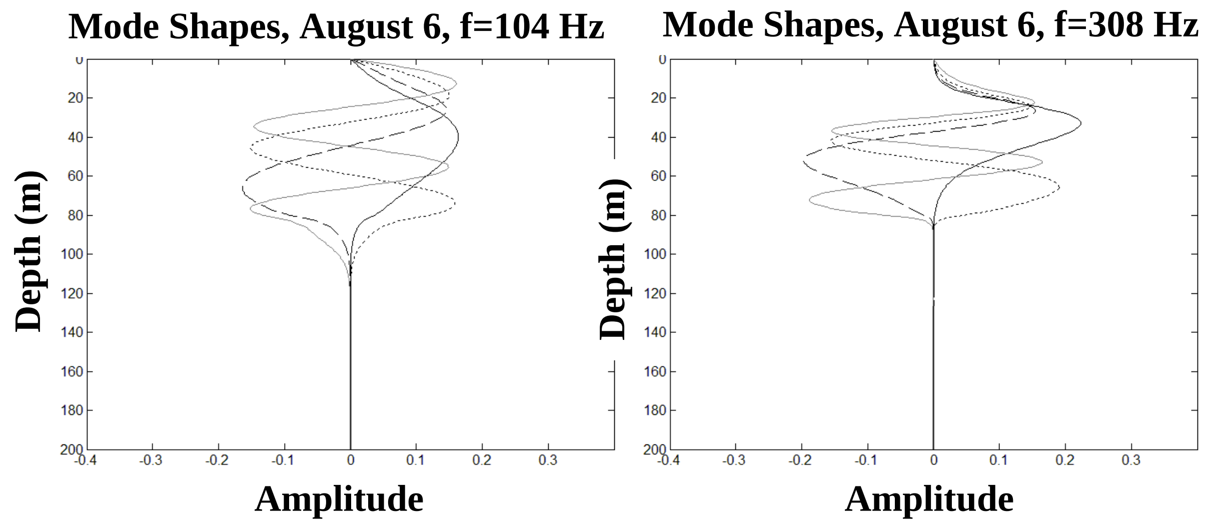


Scattering function after mode filtering

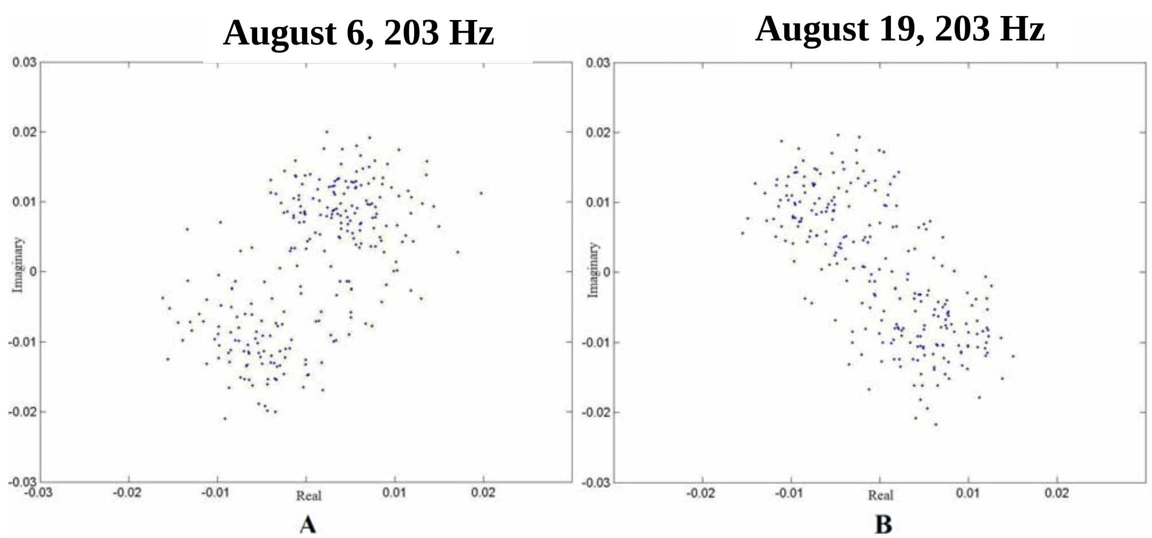


August 6

MODE FILTERING



Acoustic modes for different frequencies



Signal complex envelops constellations at the output of mode filters

DATA RECOVERY

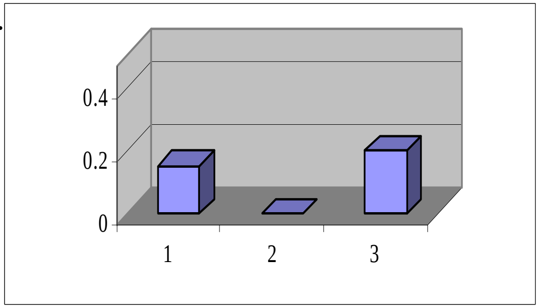
Decoder: 3 taps ML equalizer.

August 6 2006

First mode   BER= 1.5e-01

Second mode BER = 0

Third mode   BER = 0.2.

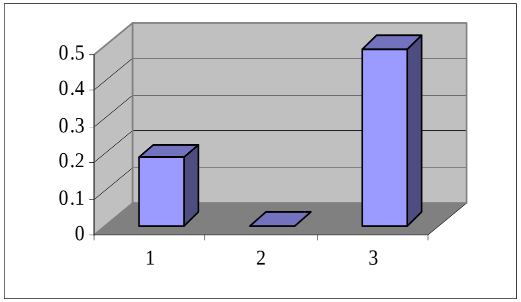


August19 2006

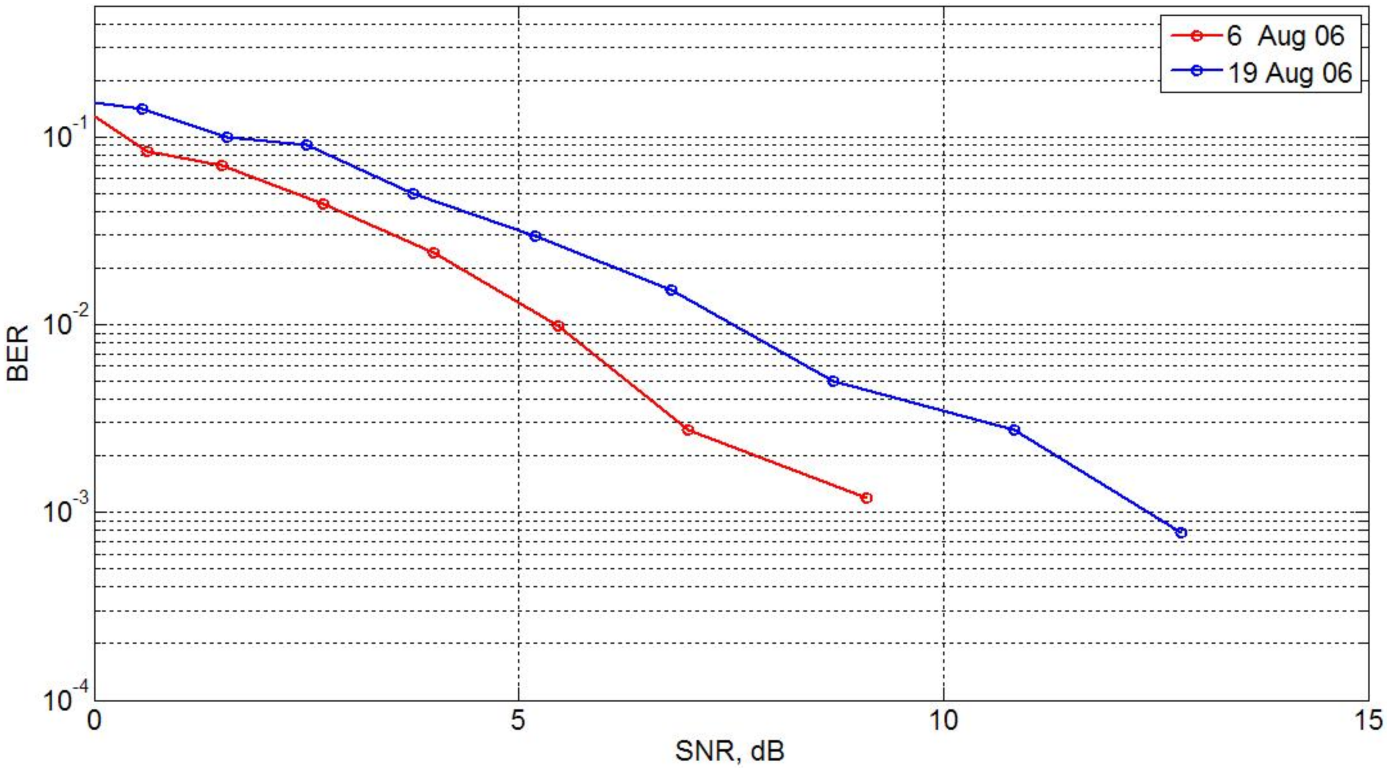
First   BER = 1.9e-01

Second   BER = 0

Third   BER = 4.9e-01



In both cases the second mode filtering gave the best reception quality with no errors



CONCLUSION

The data processing shows that even in a very complicated environment with strong internal wave solitons the acoustical energy is concentrated in a small number of the first acoustical modes. A receiver can estimate mode-time intensity distribution and use a signal from a more intensive mode (or a few of them) for demodulation. A very high quality data transmission can be achieved for a range of approximately 20 km.

Acknowledgments

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