calculate the inverse matrix which is necessary for many complexities. The number of bits which describes processing data is decided through the integer simulation. The performances of the proposed algorithm are evaluated via computer simulation in Rayleigh fading channel environment. And it is implemented as VHDL(VHSIC Hardware Description Language) to evaluate the real time processing.

68. 다중 음원 환경에서 수도형 거리추정 기법의 성능 개선에 관한 연구

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Line arrays are widely used in underwater acoustics to measure the spatial field of propagating acoustic waves and they have been investigated in the past to increase signal gain and improve angular resolution. Single-line receiver, however, requires a long horizontal aperture in order to achieve high signal gain and angular resolution; its performance may be substantially degraded by the reduced coherence lengths associated with bottom interaction in shallow water. And single-line receiver contains an ambiguity on conjugate bearings, because of lacking aperture in transverse direction. To solve the various limitations of single-line receiver the twin-line or multi-line array has been studied. Twin-line horizontal array is capable of resolving the left-right (port-starboard) bearing ambiguity inherent in single-line systems and providing high signal gain with a shorter aperture. So the drawback of twin-line array is to show lower angular resolution than single-line array, due to the short aperture size. Also, it has not been researched about array syntheses for twin-line.

In this thesis a synthetic aperture processing technique for twin-line array is proposed. The proposed method has improved SNR and extended the physical length of real line arrays by using successive measurements in space and time domain. It would be assumed that the each array moves along a straight-line course without acceleration and the received signals are coherent over the measured interval. The synthetic aperture method performs coherent processing of sub-aperture signals at successive time intervals in the beam domain via FFT transforms and is called FFT synthetic aperture (FFTSA). So the proposed method has the more improved angular resolution and the left-right bearing discrimination capability than the previous single-line or twin-line array.

Simulation results show the performance of the proposed method. To be real-time processing in real underwater environments, the proposed algorithms are implemented as DSP processor, TMS320C6711 made in Texas Instrument. The thesis presents the effectiveness of the proposed method from the implemented system. As a result, the average side lobe level

and the 3dB width were about -15.8dB and 23 in case of no synthesis. However, the proposed method by 5 syntheses showed about 22.6dB and 3, respectively.

69. 다운믹싱에 강한 디지털 오디오 워터마킹 기법에 관한 연구

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In recent year, the use of digital multimedia content has increased explosively because of rapid progress of digital and network technologies. However, this increasing of the use of digital multimedia content were raised a problem that distribute illegal copied digital multimedia content. Therefore, it was necessary to research the copyright protection technique.

The digital watermarking is one of protection technique for the digital multimedia content, which is to embed the copyright information and additional information into the digital multimedia content. And the embedded information data is called to the watermark.

The Digital audio watermarking is to embed information into the digital audio content. There are three kinds of way for embedding the digital audio watermark signal; spread spectrum coding, echo coding, and phase coding. These embedded watermark signals must be extracted although it comes under various attacks; the A/D-D/A converting, cropping, down mixing, low pass filtering, and so on.

In this thesis, we suggest the new robust watermarking technique against the down mixing attack which is satisfied with SDMI(Secured Digital Music Initiative) the Phase II Screening. The proposed watermarking algorithms are the watermark embedding and extracting algorithm in multi-channel audio data and the extracting algorithm even though multi-channel audio data was down-mixed. The proposed embedding and extracting algorithms have high information embedding efficiency as embedding PN code in each channel using interleave sequence. Also, we propose the algorithm that is able to return to original watermark signals perfectly when the multi-channel audio data applied the proposed algorithm is attacked down-mix.