5-5 Research and Development of Surrounding Computing Technology

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For comfortable information networking, it is necessary to provision variety of services for responding the requirements and to flexibly use of the distributed resources. In the “ubiquitous” environment, the distributed processing is natural to push data for real-time application. The purpose of this research is to establish the “surrounding computing technology” which is evolution of the ubiquitous environment.

In this paper, an embedded data-driven firewall processor and a signal processing method that is suitable for an information reproduction are proposed.

Keywords
Ubiquitous environment, Data-driven chip-multiprocessor, RFID, Sound field reproduction

1 Introduction

There is no need for centralized management of data in a ubiquitous environment, which is better suited for distributed processing. The objective of this R&D is to establish a surrounding computing environment, the next step in the evolution of the ubiquitous environment, in which we will be able to make use of networked computing and database resources freely without the need to be conscious of network or terminal functions.

To make effective use of information transferred through networks, all of the elements of information distributed over these networks must be organically linked, including automatic transmission processing. We are working to develop technologies to transfer and provide information in a surrounding computing environment, adding value to this information in the process. To transfer this value-added information, we must ensure that data will be sent, received and reproduced in real time, regardless of the type of data, communications media, or other factors. To this end, we are currently working on R&D of an integrated signal-processing system that can perform high-speed, highly efficient coding and that is capable of conveying information in various forms—such as image and sound data.

First, we will discuss the use of a data-driven processor in network processing aimed at flexible transmission processing in a surrounding computing environment. We will then discuss the transmission performance of an RFID reader/writer used to store information from various sensors, which is essential to the smooth execution of surrounding computing in a ubiquitous environment. While large amounts of information are transferred through ultrahigh-speed and high-capacity networks such as the JGNII, the provision of such information in a ubiquitous environment requires that this data be conveyed to numerous ubiquitous terminal devices. We thus propose a method for the effective reproduction of a sound field using such devices.

2 Network processing using DDNP

Due to recent advancements in optical transmission technologies—such as optical amplification and dense wavelength division multiplexing (DWDM)—the creation of an environment in which any form of information can be freely transmitted is now a distinct possibility. When working to provide new services flexibly over such information networks, one of the major challenges lies in developing an ultrahigh-speed programmable router chip that will accept various types of packet flows and provide suitable processing capacity according to actual link speed. A network processor unit (NPU)—a programmable high-speed packet processor—has been recently developed to meet such needs in many cases.

We are currently conducting research on the basic configuration and applications of a data-driven network processor (DDNP), a type of data-driven chip multiprocessor optimized for networks; this will be a key device in any future surrounding computing environment. The DDNP has a self-timed circuit to provide outstanding power savings and parallel processing performance.

2.1 Basic configuration of DDNP

The DDNP is a data-driven chip multiprocessor featuring an instruction set for processing of IPv4 and IPv6 multiprotocol packets on ATM or other networks.

The DDNP has a multiprocessor structure that consists of five integral arithmetic nanoprocessors mainly used for 32-bit operations, three SIMD arithmetic nanoprocessors for 8- and 16-bit operations, and two functional memory nanoprocessors for external memory access. These nanoprocessors are interconnected through an on-chip packet router. The DDNP contains instructions for CRC arithmetic, wraparound addition, and other octet-based arithmetic operations intended specifically for multi-protocol processing.

The DDNP chip is designed based on 0.18-mm CMOS technology. This chip has been verified to provide 7.5-MPPS processing, using both IPv4 and IPv6 packets[1].

2.2 Application of DDNP to add-in firewalls

With the recent widespread use of mobile devices such as cellular phones and notebook PCs, there has been an increase in demand for personal firewalls in addition to network firewalls. However, most personal firewalls are software-based and are thus liable to break down if the terminal operating system becomes infected with a virus. To address this problem, we have been conducting research on the use of the DDNP in constructing a hardware add-in firewall processor that can work independently of the operating system[2][3]. Figure 1 shows an overview of this add-in firewall.

To date, we have implemented an extended DDNP processor on FPGA, and measured the performance of a single processor through static filtering, dynamic filtering, stateful packet filtering (SPI), and URL filtering of layer-4 (TCP) packets. Table 1 shows the measurement results. As indicated by these results, it seems possible to implement firewall functions at speeds of over 100 Mbps even with a single processor[3]. As a next step, we plan to

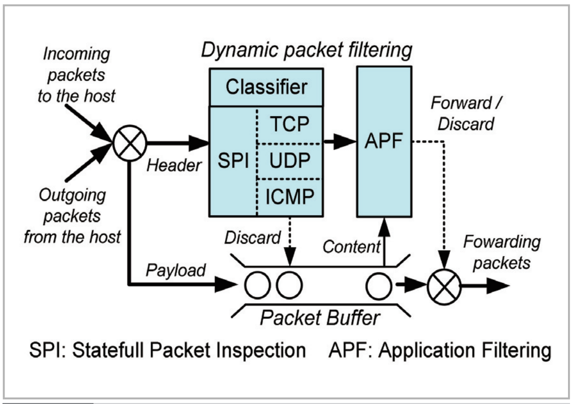

Fig.1 Overview of add-in firewall functions

Table 1 DDNP filtering performance

	Static Filtering	SPI	URL Filtering
Throughput [IP Packets/sec]	3.5M−5.0M	329K	9.1K−1.5M
Program size [DDNP Nodes]	16	443	1027

conduct research on methods of increasing the speed of content filtering; this will be essential in the safe and smooth operation of the surrounding computing environment.

3 Estimation of accuracy of RFID reader/writer

An RFID tag is a small wireless IC chip that sends information stored in its internal memory at the request of a reader/writer (interrogator)[4]. A passive RFID chip, which does not require batteries, obtains the necessary power by rectifying carrier waves sent from the reader/writer. In this section, we will estimate the accuracy of reading/writing as well as the readable/writable range when a patch antenna (micro-strip antenna) is used in both the IC chip and the reader/writer.

3.1 Four-element circular array antenna

An array antenna consists of several small antenna elements that function as a single antenna. This antenna is designed to receive the desired radio waves through the adjustment of each element in accordance with the phase and amplitude of the received signals. We used a four-element circular array antenna in this research (Fig.2).

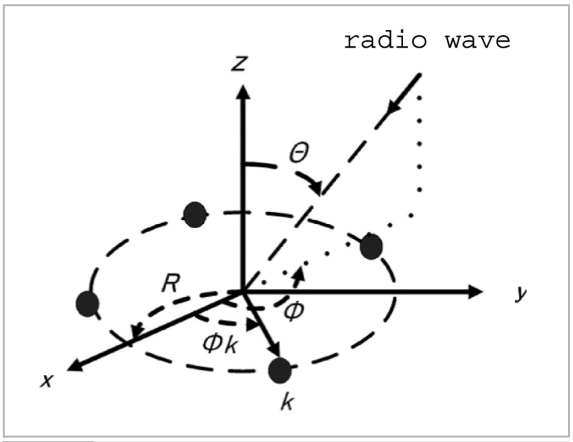

Fig.2 Circular array antenna

If antenna elements K are spaced equally around the circumference of a circle with radius R, the placement angle of a k-th element (k = 0, 1…K-1) relative to the x-axis is expressed as $\phi_k = 2k\pi/K$ [rad]. If the element is turned $\pi/4$ [rad] from its initial angle, the placement angle will be expressed as $\phi_k = 2k\pi/K + \pi/4$. In this case, the directivity of the circular array antenna is expressed by the following equation[5]:

$$F(\phi,\theta) = \sum_{k=0}^{K-1} A_k e^{j[\alpha_k + \frac{2\pi}{\lambda} R\cos(\phi-\phi_k)\sin\theta]}$$

where $A_k e^{j\alpha_k}$ is the complex weight of a k-th element; A_k is an amplitude coefficient; and ϕ_k is a phase coefficient. If the main beam is directed at an angle (ϕ_0, θ_0) in three-dimensional space, the directivity is expressed by the following equation:

$$\alpha_k = -\frac{2\pi}{\lambda} R\cos(\psi_0 - \phi_k)\sin\theta_0$$

Figure 3 shows an example of directivity of a four-element circular array antenna. Note that antenna elements used in this example are omnidirectional.

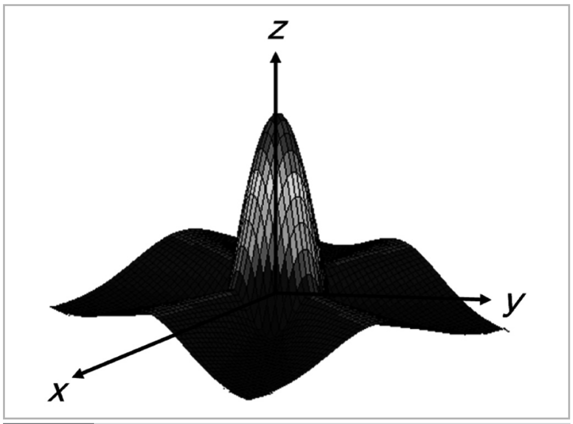

Fig.3 Example of directivity of four-element circular array antenna

3.2 Patch antenna

In this research, we used a patch antenna in both the IC chip and the RFID reader/writer[6]. The dimensions of this antenna are $2a = 2b = \lambda_g/2$ (λ_g is the wavelength within the antenna). The electric fields on the antenna's xy (ϕ= 0) and yz (ϕ= π/2) planes are expressed by the following equations:

xz plane (ϕ= 0): $E_0 = -E_0 \cos\left(\frac{2\pi}{\lambda}\sin\theta\right)$,

$$E_\phi = 0$$

yz plane (ϕ= π/2):

$$E_\phi = E_0 \frac{\sin\left(\frac{2\pi}{\lambda}\sin\theta\right)\cos\theta}{\frac{2\pi}{\lambda}\sin\theta}, \quad E_\theta = 0$$

3.3 Calculation results

We studied the relationship between the placement of the IC chip and the accuracy of the reader/writer. We calculated successful read rates when the IC chip is moved away from the reader (array antenna) in the z-axial direction, and when rotated on the reader's xz plane. Figure 4 shows the relationship between the distance and the rotation angle of the IC chip with a successful read rate of 80%. We compared three types of reader antenna: (1) a single-element patch antenna; (2) a four-element circular array consisting of the elements in (1); and (3) a four-element circular array consisting of the arrays in (2) (16 elements in total). As shown in the figure, using the reader described in (3) will allow for an expanded range for chip distance while maintaining a read rate of 80%.

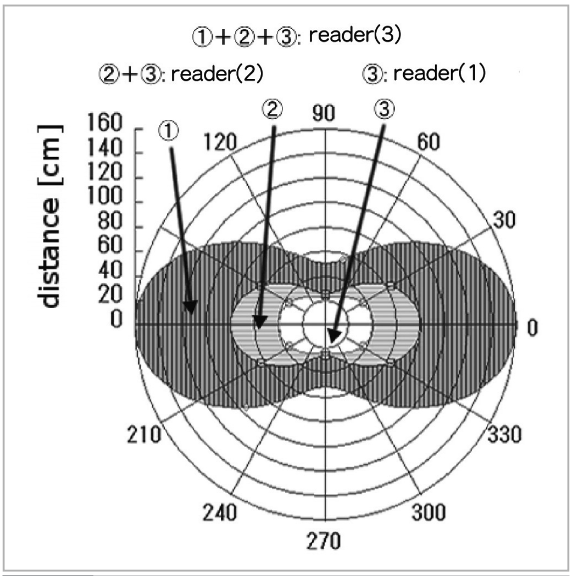

Fig.4 Estimated distance between reader and IC chip when read rate is 80% or higher (Relationship between angle and distance of IC chip)

4 Sound Field Reproduction System

In this section, we will propose a system for reproducing stereo acoustic information that is the simplest in structure and easy to implement. To reproduce a certain sound field, it is necessary to cancel out the effects of the transfer characteristics from the loudspeaker (secondary sound source) to the microphone (control point), i.e. to approximate a characteristic that is the inverse of this transfer characteristic. To date, various inverse-characteristic approximation methods have been reviewed based on system-related methodologies. A multi-channel/multi-point control system based on the MINT (multi-input/output inverse theorem) is now considered to be the most effective method[7]. If the number of control points is M, this method enables sound reproduction using M +1 secondary sound sources to approximate the inverse characteristic and to eliminate crosstalk. However, such a control system becomes complex with use of multiple loudspeakers. We will thus propose a new control system consisting of two control points and two secondary sound sources.

4.1 Multi-input signal correction system

Figure 5 shows the configuration of our proposed two-channel/two-point control system (multi-input signal correction system). An adaptive filter H_{11} approximates a characteristic inverse to G_{11} by calculating the output error from a right-channel input signal $d_1(t)$ and a filtered signal $s_{11}(t)$. Since there is a strong correlation between crosstalk transfer characteristics G_{12} and G_{21}, the filter coefficient obtained by H_{21} can be used as a filter to estimate the characteristic G_{12}. By giving this as the right-channel correction filter coefficient c_1, the crosstalk component passing through G_{12} can be made to approximate the desired left-channel signal $x_2(t)$. Therefore, by obtaining a right-channel correction filter coefficient that is a combination of the filter coefficients obtained by H_{11} and H_{21}, it is possible to design a correction filter capable of reducing the effects of crosstalk. The right- and left-channel correction filter coefficients are expressed as:

$$\boldsymbol{c}_{1,L}(t+1)=w_1\boldsymbol{h}_{11,L}(t)+(1-w_1)\boldsymbol{h}_{21,L}(t)$$
$$\boldsymbol{c}_{2,L}(t+1)=w_2\boldsymbol{h}_{22,L}(t)+(1-w_2)\boldsymbol{h}_{12,L}(t)$$

where w_i is a weighted parameter ($0 < w_i \leq 1$) for the first and second terms of the equation. To update the filter coefficient of an adaptive filter H_{ij}, we apply a learning identification method[8] expressed as:

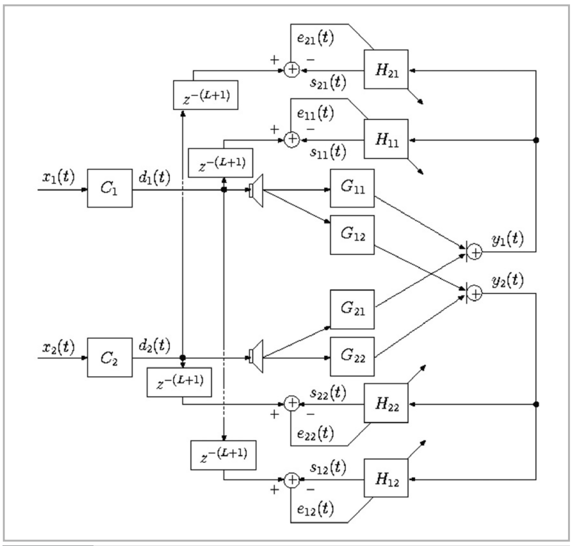

Fig.5 Configuration of multi-input signal correction system

$$\boldsymbol{h}_{ij,L}(t+1)=\boldsymbol{h}_{ij,L}(t)+\alpha\frac{\boldsymbol{y}_{j,L}(t)}{\|\boldsymbol{y}_{j,L}(t)\|^2}e_{ij}(t)$$

where α is the step gain and e_{ij} is the output error of the filter.

4.2 Calculation of output error based on auditory characteristics

When calculating output error, we assign weights based on human auditory characteristics to reduce unnecessary frequency bands and to improve calculation efficiency. For these human auditory characteristics, we use the so-called "A-weighted sound pressure level",[9] which is used as a measure of noise. A weighted function is calculated for each frequency band using the following equation:

$$\phi(k)=\frac{\mathrm{Res}(k)+|\min\mathrm{Res}|}{|\min\mathrm{Res}|}$$

where Res(k) and minRes are frequency responses based on the A characteristics and the minimum value, respectively. We multiply the input signal $d_i(t)$ and the adaptive filter's output signal $s_{ij}(t)$ by $\phi(k)$ in the frequency domain and obtain the difference between the two values in the time domain as a new output error e_{ij}.

4.3 Computer simulations

We performed computer simulations to verify the effectiveness of our proposed method, comparing convergence characteristics using different values for step gain a and for weighted parameter wi.

(1) Simulation 1

As the desired signal $x_i(t)$, an audio signal is sampled from adult male voice at 8 kHz. The adaptive filter's impulse response length L is set to 512. The transfer function for use in this simulation is determined by actual measurement of initial signals (white noise) and observed signals in a reverberant room with three different layout patterns of loudspeakers and microphones. Convergence characteristics are then compared using different values for step gain a and for weighted parameter wi, without assigning weights, based on the auditory characteristics. Reproducibility of sound is assessed (SNR) relative to the original audio as follows:

Table 2 Degree of improvement, parameters, and cross-correlation coefficients

	α	w_1	w_2	r_N	Improvement[dB]
Pattern 1	0.05	0.35	0.25	0.73	6.23
Pattern 2	0.05	0.35	0.50	0.55	5.40
Pattern 3	0.05	0.45	0.85	0.45	5.13

$$\mathrm{SNR} = 10\log_{10}\frac{E[x_j^2(t)]}{E[y_j^2(t) - x_j^2(t)]} \quad (1)$$

where $E[\ldots]$ is the expected value. SNR is calculated without correction to the desired right- and left-channel signals, and SNR is calculated using our proposed method. The difference between the two values is then determined as the degree of improvement.

Table 2 shows the conditions in which the average degree of improvement in the right and left channels is greatest for each of the three patterns. Although there is considerable variation among the weighted parameters, the step gain α is always approximately 0.05 when the degree of improvement is largest. r_N is a coefficient of cross-correlation between two crosstalk components. The larger the value of r_N,, the larger the degree of improvement.

(2) Simulation 2

Weights are assigned based on auditory characteristics when calculating output error e_{ij}, and acoustic reproduction characteristics are compared where the degree of improvement becomes largest. Weight $\phi(k)$ is assigned in the frequency bands of the desired signal $x_i(t)$ and of the observed signal $y_j(t)$, and the degree of improvement is calculated from SNR in equation (1) above.

In Fig.6, degrees of improvement are compared for pattern 1 using the following values: α = 0.05, w_1 = 0.35, w_2 = 0.25. These results indicate that an improvement in speed of approximately 0.63 seconds is achieved by assignment of weights based on auditory characteristics.

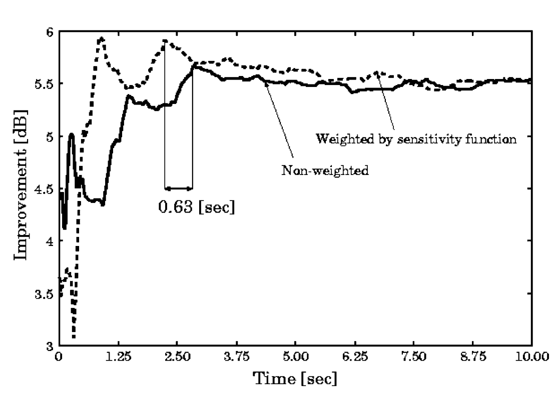

Fig.6 Comparison of acoustic reproduction characteristics

5 Conclusions

In this paper we have described a network processor based on a data-driven processor, and discussed the transmission performance of an RFID reader/writer. We have also proposed a method for the effective reproduction of an acoustic space on ubiquitous terminal devices.

These research results can be used as constituent technologies in the establishment of a surrounding computing environment. Based on these results, we intend to create a surrounding computing environment that will enable the effective use of data and resources distributed over networks.

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