Laboratory Introduction
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Annual Report 2007
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He received the Paper Award from the IEICE in 1995 and

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Assistant Professor Satoshi Suyama
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Suzuki Fukawa Laboratory

Recent Our Research Outline and Topics

Research Outline

Transmission System
- MIMO
- OFDM
  - Turbo equalizer to cancel ISI and ICI for large delay spread
- Channel estimation for
  - Preamble-type: Radio LAN
  - Scattered-type pilot schemes: Digital Terrestrial TV
- Efficient pilot scheme for very short radio packet
- Phase noise compensation processing for millimeter-wave communication

Multiple Access
- Access Scheme
- Multiuser Detection

In-house Design & Implementation of Mobile Communication System Simulators
- Gaussian Multicarrier (GMC)
- Bandpass Over-Sampling
- SSB
- FPGA On-board System Simulators
- 4 x 4 MIMO Fading Simulators

Research Topics of Recent Five Years Period

Transmission System
- OFDM
  - Turbo equalizer to cancel ISI and ICI for large delay spread
- Channel estimation for
  - Preamble-type: Radio LAN
  - Scattered-type pilot schemes: Digital Terrestrial TV
- Efficient pilot scheme for very short radio packet
- Phase noise compensation processing for millimeter-wave communication

- MIMO
  - Detection and CSI estimation
  - Sub-optimal ML
    - EM algorithm with factor graphs
  - MMSE detection avoiding noise enhancement
  - Adaptive blind MIMO detection for heterogeneous streams
  - Minimum BER design of precoding with CSI
    - Rayleigh fading channel environment
    - Interference dominant environment
    - Pre-FFT and post-FFT enhancement schemes
  - Frame synchronization

- MIMO-OFDM
  - BER improvement
    - Subcarrier phase hopping (SPH)
    - Soft decision-directed channel estimation (SDCE)
  - PAPR reduction
    - Enhanced selected mapping (ESLM)
  - Partial transmit sequence (PTS)
  - Joint processing PAPR reduction and BER improvement
    - SPH-SLM
    - Eigen-mode transmission

Multiple Access
- Access scheme
  - IDMA
  - Multiuser MIMO
  - Random Packet Collision Resolution
- Multiuser detection
  - Interference canceller
    - Systolic array scheme
    - Array combining

Modulation and Demodulation for Cognitive Radio
- Gaussian Multicarrier (GMC)
- Bandpass over-sampling
- SSB

In-house Simulator Design and Implementation
- FPGA on-board system simulators
  - Simple MIMO (Vertex II)
  - Large scale MIMO (Vertex IV)
  - VHDL design from scratch
- 4x4 MIMO fading simulators

Our laboratory has researched the signal processing for mobile communications. Recently, we have especially focused on the transmission systems, such as OFDM, MIMO, and MIMO-OFDM, and multiuser detection for various multiple access schemes. Additionally, modulation and demodulation schemes, such as Gaussian multicarrier, bandpass oversampling, and SSB, have been researched for cognitive and software radio systems. Moreover, the transmission systems with these research topics and fading simulator have been implemented on FPGA-board simulator to verify the real time processing. These are summarized in the upper illustration and the detailed research topics of recent five years period are enumerated in the lower part. This report introduces some topics from them in the following sections.

**Linear Precoding Technique for Improving Detection Reliability of MIMO-OFDM System Employing MLD**[8]

MIMO-OFDM has recently attracted much attention because it can increase channel capacity by spatially multiplexing independent data streams. If the channel state information (CSI) is available at the transmitter, a linear precoding technique can exploit the channel more efficiently. This report presents a new linear precoding technique that improves the detection reliability of the MIMO-OFDM system employing MLD.

Fig. 1 shows the received signal replicas generated by MLD of a single-cell single-user MIMO-OFDM system with two QPSK modulated data streams, where (a) does not employ any precoder and (b) employs a minimum BER (MBER) precoder which will be explained later. Assume that b and r denote the transmitted and received signal points, respectively. Fig. 1 (a) shows that r is displaced from b because of the additive white Gaussian noise (AWGN). Since the MLD makes a decision in favor of the signal point locating closest to the received signal, the receiver decides that c is transmitted.
Therefore, the detection error occurs. In order to reduce the amount of detection error and to improve the reliability of MLD, the MBER precoder optimizes the Euclidean distances of the received signal replicas in Fig. 1 (a) and the result is shown in Fig. 1 (b). In Fig. 1 (b), the MLD can make a correct decision with these new replicas.

To implement the above concept of the MBER precoder, the proposed method calculates a theoretical BER upper bound of MLD by using the pairwise error probability (PEP). Then, the steepest descent algorithm is employed to estimate the precoder that can minimize this bound. Fig. 2 shows the average BER performances of the nonprecoded, the conventional MBER precoded, and the proposed MBER precoded schemes. Two conventional MBER precoding methods were investigated including the minimum distance-based precoder and the power allocation MBER precoder. Fig. 2 shows that the proposed method gives significant improvement over the conventional schemes.

The framework of the MBER criterion precoding described above can be extended to the multi-cell environment, where one mobile station is communicating with one base station and other base stations are generating interferences. In this situation, the condition for the detection error of MLD does not only depend on the AWGN but also the interferences from other base stations. Under the assumption that the interferences can be reliably estimated, the theoretical BER upper bound of MLD can be calculated again by using the PEP. Then, the precoding parameters can be optimized in order to minimize this bound and the result is expected to show better improvement in the average BER than the precoder that only considers the AWGN for the optimization.
Scattered Pilot OFDM Receiver Employing Iterative Detection and Channel Estimation in Fast Fading Environments[22]

OFDM is one of the most promising techniques for high-speed data transmission in mobile communications, the wireless LANs, and digital terrestrial broadcasting such as ISDB-T in Japan and DVB in Europe. ISDB-T can efficiently reuse the frequency resource by employing single frequency network (SFN) to extend a service area. Since signals from several broadcast stations can arrive in SFN, delay time of a propagation path may reach as long as the guard interval (GI). Therefore, an OFDM receiver for the mobile reception service of ISDB-T which adopts the scattered-pilot (SP) needs to estimate a channel impulse response including long delay propagation paths with the sufficient accuracy and the tracking performance. The accurate channel estimators, which estimate the channel impulse response by using both SP of some symbols and the log-likelihood radio (LLR) of the coded bits from the decoder, have been already proposed. However, in the OFDM transmission with a large number of subcarriers, they cannot achieve sufficient tracking performance because the number of estimated parameters increases. To solve this problem, we have proposed an iterative SP-OFDM receiver that extracts only dominant paths by LLR and SP and that conducts soft decision-directed channel estimation (SDCE) by the LMS algorithm which selects and updates only the taps corresponding to the extracted paths.

Fig. 1 shows the configuration of a proposed OFDM receiver. It consists of an initial channel estimator (ICE), a coherent detector, a MAP decoder, a tap selector, and SDCE, and it operates with the initial and the iterative processing. In the iterative processing, the tap selector first estimates the channel frequency response by inverse-modulation of the received signal with SP and the mean of the modulation signal, and it calculates the approximately estimated channel impulse response. Next, the tap selector extracts only dominant paths from the estimated impulse response and selects the corresponding delay time which SDCE employs for the LMS algorithm. Moreover, SDCE estimates the impulse response by using OFDM signal replicas which are generated from SP and the means of the modulation signal as the input signal for the LMS algorithm.
Then, the coherent detection, and MAP decoding are conducted. The receiver iterates a series of the processing with tap selection, SDCE, the coherent detection, and MAP decoding, until the number of iterations exceeds a predetermined threshold or decision errors are no longer detected.

**Fig. 2 Average BER performance**

Fig. 2 shows average BER performance of the iterative SP-OFDM receiver with 4 iterations, when the maximum Doppler frequency $f_D$ is equal to 80 Hz ($f_D T_S = 0.1$). In this situation, the speed of the mobile terminal becomes 160 km/h at the ISDB-T carrier frequency of 500 MHz. SDCE-CD with the adaptive tap selection and without the tap selection were also evaluated. For comparison, BER of only the initial processing is plotted. It can be seen that the conventional SDCE-CD without TS cannot track the channel fluctuation because of a large number of the parameters. Therefore, it cannot improve the performance, while the proposed method with TS can achieve an average BER of $10^{-4}$ at the average $E_b/N_0 = 13$ dB, and gain 6 dB over SDCE-CD without TS.

**Adaptive Blind Detection for Heterogeneous Stream MIMO Systems**[16],[23]

MIMO mobile communication system have recently attracted much attention, because it can realize high data-rate transmission without expanding the frequency band. Conventional MIMO systems utilize training sequences in order to perform stream separation, signal detection, and channel estimation. Meanwhile, the blind detection that does not need any training sequences are preferable from the viewpoints of the bandwidth efficiency and channel tracking. However, the blind MIMO detection cannot principally identify the separated streams, which refers to the stream ambiguity. Although insertion of unique words resolves the ambiguity, it increases the packet overhead and reduces the bandwidth efficiency.
For solving such a problem without the unique words, we have considered a heterogeneous stream (HTS) scheme. To identify the respective streams, HTS brings in different properties such as modulation, coding, or their different parameters, which are previously known to the receiver and available for identifying the detected streams. More specifically, two effective examples of HTS are discussed: modulation type HTS (MHTS) and timing-offset type HTS (THTS). MHTS utilizes modulation sets having the same bandwidth but different constellations for respective streams, as shown in Fig 1 (a), while THTS utilizes the modulation sets having the same constellation but the different transmission timing, as shown in Fig 1 (b).

For the blind detection, the receiver employs maximum likelihood sequence estimation with per-survivor channel estimation which can track the fast fading. The receiver also employs fractional sampling and the channel estimation is updated in every fractional timing for further improvement of the tracking. Additionally the fractional sampling also enhances the robustness of the signal detection with Viterbi algorithm for THTS because each stream can be received in each timing.
Computer simulations for 2 x 2 MIMO systems with the HTS schemes have shown that the system can achieve superior BER performance with $E_b/N_0$ degradation of 1.0 dB in THTS and 2.5 dB in MHTS compared with the ideal maximum likelihood detection at average BER $10^{-3}$ over a frequency-flat Rayleigh fading channel as shown in Fig.2 (a). The detector shows superior tracking performance up to $f_{DT}=2.0 \times 10^2$ as shown in Fig.2 (b).

**430 Mbps Reduced-PAPR MIMO-OFDM Transceiver on FPGA-Board Simulator [9],[19]**

In space division multiplexing (SDM) of the coded MIMO-OFDM, the spatial spreading and subcarrier phase hopping for SDM (SPH-SDM) have been proposed in order to increase the frequency diversity gain. These techniques intentionally reduce the channel correlation between the subcarriers by rotating the phase of each subcarrier individually. However, they cause high peak-to-average power ratio (PAPR) like the normal MIMO-OFDM, and thus degrade an efficiency of the transmitter power amplifier. To overcome this impairment, SPH-SLM, which combines SPH-SDM with the selected mapping (SLM) for PAPR reduction, has been proposed. Since SPH-SLM can not only improve the transmission performance but also reduce PAPR, it is one of the most promising techniques for the future mobile communications. This report presents a 432 Mbps 2 x 2 MIMO-OFDM transceiver employing SPH-SLM implemented on an FPGA-board simulator.

A block diagram of the implemented SPH-SLM transmitter is shown in Fig.1. First, information bits are divided into all the streams. Next, each stream is encoded, and the transmitter maps them into a modulation signal at each subcarrier. Then, it carries out the subcarrier phase hopping (SPH) to the modulation signals of all streams.

SPH performs the different phase shifts for the modulation signals according to the indices of subcarriers, transmit antennas, and phase patterns. A phase pattern controller selects the optimum phase pattern that can minimize the peak power of every symbol. A phase pattern controller, shown in Fig.2, consists of SPH units, IFFTs, and peak detectors, which are parallely
The complementary CDF for the threshold of PAPR was measured by the simulator and was depicted in Fig.4. The results obtained by the floating-point computer simulations are also plotted for comparison. Number of the phase patterns was changed to 1, 4, 8, 16. It can be seen that with 16 phase patterns, it can gain 3.4 dB at CCDF $10^{-3}$ in comparison with the normal MIMO-OFDM transmitter, and that it can achieve PAPR of less than 7 dB.

Fig4. PAPR performance

The SPH-SLM transmitter and receiver were implemented on the FPGA-board simulator which has six FPGA chips of Xilinx Virtex-II Pro XC2VP70 as shown in Fig.3. Overall configuration of the simulator consists of the fading generator, noise generator, and transmission error counter in addition to the SPH-SLM transceiver. Although the transmission parameters for OFDM mainly follow IEEE 802.11a standards, the bandwidth was set to 80 MHz. The simulator can support QPSK, 16QAM, and 64QAM and employs the convolutional code with the coding rate of 1/2, which is punctured to 2/3 or 3/4. Therefore, 2 x 2 MIMO using 64QAM with code rate 3/4, the simulator can achieve a maximum bit-rate of 432 Mbps.
Professor Jun-ichi TAKADA

was born in Tokyo, Japan, in 1964. He received the B.E., M.E., and D.E. degrees from the Tokyo Institute of Technology, Tokyo, Japan, in 1987, 1989, and 1992 respectively. From 1992 to 1994, he was a Research Associate at Chiba University, Chiba, Japan. From 1994 to 2006, he was an Associate Professor at the Tokyo Institute of Technology before becoming a Professor. He has been participating in European COST action 2100 “Toward mobile broadband multimedia networks.” His current research interests are wireless propagation and channel modeling, array signal processing, Cognitive radio, and application of wireless communication and information technology for regional/rural development. Dr. Takada is a member of IEEE, IEICE, Applied Computational Electromagnetics Society (ACES), and ECTI Association, Thailand.

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Asst. Prof. Kim was born in Seoul, Republic of Korea. He received the B.S. degree in Electrical Engineering from Hanyang University, Seoul, Korea, M.E. and Ph.D. degrees in Division of Electrical and Computer Engineering, Yokohama National University (YNU), Japan in 1999, 2002 and 2005, respectively. He was with a startup company from 2005 and has experienced H/W and S/W development of various embedded system. He was also with YNU as a postdoctoral research fellow shortly in 2006. He joined Tokyo Institute of Technology (Tokyo Tech) as an assistant professor from July 2007. His research interests include digital signal processor implementation, radio propagation measurement, array signal processing, smart antenna system, software defined radio / cognitive radio. He is a member of IEEE and IEICE.
Paper Awards of Takada Laboratory

Gilbert won First Prize for Best Student Scientific Paper at World Congress on ITS

Panarat won Student Paper Award at APMC 2007

Member of Takada Laboratory
Real-Time Propagation Measurement System for Electric Toll Collecting (ETC) System

Introduction

In the early deployment of electric toll collecting (ETC) system, multipath interference has caused the malfunction of the system. Therefore, radio absorbers are installed in the toll gate to suppress the scattering. This paper presents a novel radio propagation measurement system to identify the individual scattering object and the power intensity of the ETC gate in real time.

<table>
<thead>
<tr>
<th>Table 1: System specification</th>
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<tbody>
<tr>
<td><strong>Signal</strong></td>
</tr>
<tr>
<td><strong>Modulation</strong></td>
</tr>
<tr>
<td><strong>Carrier frequency</strong></td>
</tr>
<tr>
<td><strong>Rx power range</strong></td>
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<tr>
<td><strong>Array geometry</strong></td>
</tr>
<tr>
<td><strong>Element spacing</strong></td>
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<tr>
<td><strong>Antennas element</strong></td>
</tr>
<tr>
<td><strong>RF frequency</strong></td>
</tr>
<tr>
<td><strong>Bandwidth</strong></td>
</tr>
<tr>
<td><strong>Sampling frequency</strong></td>
</tr>
<tr>
<td><strong>Quantization bit</strong></td>
</tr>
<tr>
<td><strong>FPGA</strong></td>
</tr>
<tr>
<td><strong>FPGA memory</strong></td>
</tr>
</tbody>
</table>

Measurement System

The signal and system specifications are presented in Table 1. The measurement system captures the data in burst-wise manner at each time frame in a moving vehicle. The vehicle loading this system usually passes through the toll gate at constant speed of 20 km/h. In the measurement, the laser sensor is used to trigger for the start and end positions of the measurement and the system captures the data at every equi-time interval along the running path until the end-trigger is detected.

In this system, DOAs are estimated with a linear antenna array with cone ambiguity. To overcome this, this paper proposes the scattering object identification method using 3-D visualization of beamforming results that include the signal power intensity at any direction along the observation points. The beamforming results is applied to a spherical beampatterns that express the corresponding power intensity by its transparency in a cone shape on its surface. It means making any parts of the surface transparent when the power is higher than a threshold value and making them colored otherwise. When producing the spherical beampattern, the threshold level should be appropriately selected. The spherical beampatterns is converted to VRML (virtual reality modeling language) with the 3-D model of the toll gate.
Results and Discussion

Figure 1 presents the measurement results on the adjacent lane in case of water sprinkler exits in ETC lane. The length of the measurement area was 13m and the transmitter antenna location in the ETC lane was x=8m from the start point of the measurement. As shown in Fig.1(a), the vehicle with this system carries out the measurement passing through the toll gates. Fig.(b) shows the contour plot of the beamforming pattern that provides the power intensity distribution. From this results, we can find large leak power about -55 dBm to the adjacent lane caused by the reflection of the water sprinkler at around x=2m and \( \theta_0 = 140 \) deg. It means that the malfunction can occur by this leaked signal because the threshold power level in the ETC receiver operation is prescribed by -70.5dBm. The results of the scattering object identification using 3-D beampatterns are shown in Fig.(c). The threshold level in producing the spherical beampattern was set by the lower bound of the ordinary receiving power, -60.5dBm. It can be seen that the water sprinkler is identified as a scattering object and the power intensity is also easily found.

Conclusion

This paper proposes a novel radio propagation measurement system useful for the ETC gates without closing the gate. The measurement can be made on the vehicle passing through the gates using real ETC signal transmitted by ETC base station. This system can identify the individual scattering object and the power intensity of the ETC gate as well as the spatial power intensity distribution in real time. Moreover, the validity of the system was verified by the field experiment.
Identification of Relatively Strong Clusters in an NLOS Scenario at a Small Urban-Macrocell Mobile Station

Introduction

The need to model the collection of multipaths (MPCs) that lie in the same angle-delay domain, or clusters, is also one of the things needed to approach the full performance advantage of Multiple-Input Multiple-Output (MIMO) systems for future cellular communications.

• Researchers have various ways of defining a cluster.
  ⇒ Cluster size & location highly depends on the physical environment.
• Current research focuses on microcells & picocells but not much on macrocells.

We present a procedure for identifying clusters around the mobile station in a small macrocell. It is based on the parameters of stronger clusters, which serve as the initial centroids. Using this method, most of the clusters were identified in an NLOS scenario. Employing this approach may aid in the identification of clusters when relatively strong paths below the path power of the centroids are included.

Channel Sounding Setup

Table 1: Medav RUSK-Fujitsu MIMO Channel Sounder

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier frequency</td>
<td>4.5 GHz</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>120 MHz</td>
</tr>
<tr>
<td>BS Antenna</td>
<td>Uniform Rectangular Array</td>
</tr>
<tr>
<td>MS Antenna</td>
<td>Stacked Uniform Circular Array</td>
</tr>
<tr>
<td>Tx Signal</td>
<td>Wideband Multicarrier Spread Spectrum</td>
</tr>
<tr>
<td>Tx Power</td>
<td>40 dBm</td>
</tr>
<tr>
<td>Maximum path delay</td>
<td>3.2 ps</td>
</tr>
</tbody>
</table>

⇒ Estimated parameters: azimuth & elevation AoA, azimuth & elevation AoD, time delay, & the amplitude of the 4 polarimetric MPCs.
• Results are independent of the antennas used.
⇒ Double-directional channel concept based measurement & estimation system.
Results and Discussions

• Paths considered: until 20dB below the normalized strongest path (0dB) ⇒ Scenario length: 2m (frame containing five snapshots) the paths within it did not change considerably.
• More clusters can be seen after we included the paths below −6dB. See Fig. 2.
• As can be seen in Fig. 1, C1 in Fig. 2 had the largest concentrations of MPCs.
• The results also show that each cluster is unique ⇒ Traditional uniform distribution of scatterers around the MS is not always appropriate.
• Overall, the identified clusters accounted about 68% of the path power with respect to the total power (including the diffuse components).

Fig. 1: Initial clusters defined with their parameters. Order of the arrow label: the mean azimuthal AoA, the mean time delay, & the mean elevation AoA of the centroids.

Fig. 2: Time delay against the azimuthal AoA. C1 refers to cluster 1 while U1 refers to unclustered path 1.

Conclusion

• Using the stronger path centroids enabled us to identify clusters when all the considered paths were included.
• But there remains some unclustered paths that may not easily be identified by the procedure.
• This work may augment the initial selection of cluster centroids in automatic clustering.
Ray Tracing Simulation of Microcell Propagation Channels

Introduction

For any wireless systems, ray tracing can be used to simulate the multipath propagation environment. But due to its geometrical optics (GO) approximation and computational limitation, error occurs depending on the environment. In this work, we first survey the range in a microcell environment where the GO can be calculated. Next we introduce

the complex radar cross section for ranges that are impossible to calculate by GO. We then perform the correct simulation by simple expansion of the ray tracing simulator.

Applicable Range of Geometrical Optics

Propagation path loss at long distances decreases on the street model (Fig.1) which is usually well observed in an urban area. The reason is that there are a lot of diffraction rays with large energy near the specular reflection. The reflection and diffraction coefficients in GO are used on the assumption that the size of reflection and diffraction edge are infinite. We therefore survey the condition of applicable range where GO can be used, by comparing GO and the physical optics (PO), which gives better accuracy but longer simulation time. In this work, we compare PO and the stationary phase method (SP) which is a good approximation of PO and is equal to GO. Note that while the computation of GO can be separated into a reflection and diffraction component, the result of PO in this model combines already both components. In this section,

multiple objects are designed on the assumption that only one plane connecting the multiple objects is allocated, and its center is the reflection point. PO diffraction is calculated as the difference between PO and SP's reflection component. The diffraction of GO is SP's diffraction. We note that PO's diffraction and SP's diffraction are different within the 1st Fresnel zone. It is therefore not applicable to use GO in this range.
Introduction of Complex Radar Cross Section

The radar cross section (RCS) is introduced for the range where GO is not applicable. The scattering rays are designed using RCS instead of specular reflection and diffraction in the 1st Fresnel zone for specular reflection (Fig.3). In this work, the complex RCS is introduced in order to calculate the scattering of multiple objects. The complex RCS is based on the 3-D plane model which is not a perfect conductor.

Simulation Result

Simulation result (Fig.4) is calculated based on Fig.1. Measurement data [1] with model similar to Fig.1 is also included. In the conventional technique the received level increases at long distances. In the proposed technique, the received level decreases, and is close to the measurement data.

Conclusion

In this work, we point out the problems of the geometrical optics approximation and examine the range that geometrical optics cannot apply. Next we introduce the complex RCS in ray tracing, and verified that the proposed technique is closer to measurement data than the conventional technique.

References

Performance Evaluation of Cyclostationary Detector for Cognitive Radio System

Introduction

Spectrum sensing is one of the most important techniques for cognitive radio systems because it can recognize the presence or the absence of primary signals. In this report, two modified generalized likelihood ratio test (GLRT) for spectrum sensing based on cyclostationarity are presented. One is based on sensing at multiple cyclic frequencies and the other is based on cooperation sensing among cognitive users.

Cyclostationarity Detector

To detect whether or not a primary signal $x(t)$ is present, a row vector

$$\hat{R}_{xx}^\alpha(\tau) = \text{Re}\{R_{xx}^\alpha(\tau)\}, \text{Im}\{R_{xx}^\alpha(\tau)\}$$

is examined. In this case, $\hat{R}_{xx}^\alpha(\tau)$ the estimation of cyclic autocorrelation function (CAF) which is defined by

$$R_{xx}^\alpha(\tau) = \frac{1}{T} \sum_{t=0}^{T-1} x(t)x^*(t+\tau)e^{-j2\pi \alpha t}$$

For a given cyclic frequency $\alpha$ and delay $\tau$, the signal detection can be based on hypothesis testing problem which is stated as follows

$$H_0: \hat{R}_{xx}^\alpha(\tau) = \psi_{xx}^\alpha(\tau), \text{signal is absent}$$

$$H_1: \hat{R}_{xx}^\alpha(\tau) = \hat{R}_{xx}^\alpha(\tau) + \psi_{xx}^\alpha(\tau), \text{signal is present}$$

where $\psi_{xx}^\alpha(\tau)$ is an estimation error. For $T$ is large, it is known that $\sqrt{T}\hat{R}_{xx}^\alpha(\tau)$ is Gaussian with probability density function (pdf) $\mathcal{N}(0, \Sigma_{T\tau}^\alpha(\tau))$, where $\Sigma_{T\tau}^\alpha(\tau)$ is the estimated covariance matrix.

Based on generalized likelihood ratio test (GLRT) the statistical cyclic test can be defined as follow

$$T^\alpha(\tau) = \frac{1}{T} \sum_{t=0}^{T-1} (x(t)x^*(t+\tau)e^{-j2\pi \alpha t})^2$$

- Under $H_0$, $T^\alpha(\tau)$ converges to a central Chi-square distribution $\chi^2$
- Under $H_1$, $T^\alpha(\tau)$ converges to Gaussian distribution with pdf $\mathcal{N}(0, \Sigma_{T\tau}^\alpha(\tau))$

be determined. Next, the detection can be performed as follows

$$H_0: T^\alpha(\tau) < \gamma, \text{signal is absent}$$

$$H_1: T^\alpha(\tau) \geq \gamma, \text{signal is present}$$
Results and Discussions

The conventional cyclostationary detector detects the presence or absence of the primary signal based on a single user at a single cyclic frequency. Therefore, the performance of this detector cannot be obtained significant improvement over other existing detectors such as energy detector. In this report, the detection based on multiple cyclic frequencies and based on multiple cognitive users are presented. The detection performances are evaluated in a noisy observation while an orthogonal frequency division multiplexing (OFDM) is considered as the primary user. Each cognitive user shares the information about the cyclic test $T^x_f$ in order to find the vacant spectrum. Fig. 1 confirms that the detection performance $T^x_f$ be improved as the number of cooperative users is increased. Fig. 2 shows that the probability of detection increases as the probability of false alarm is increased.

![Fig. 1 probability of detection vs. SNR with probability of false alarm of 1%](image1)

![Fig. 2 probability of detection vs. probability of false alarm with SNR=-14 dB](image2)

Conclusion

The cyclostationary detector outperforms the detection performance in low SNR and has the desirable receiver operating characteristics. The detection performance can be improved by sensing at multiple cyclic frequencies and it has also indicated the significant gains of performance via the cooperative sensing.
Associate Professor Kei Sakaguchi

Assoc. Prof. Sakaguchi (right) was born in Osaka, Japan, on November 27, 1973. He received the B.E. degree in electrical and computer engineering from Nagoya Institute of Technology, Japan, in 1996, and the M.E. degree in information processing and the Ph.D. degrees in electrical & electronic engineering both from Tokyo Institute of Technology, Japan, in 1998 and 2006, respectively. In 2000-2008, he was an Assistant Professor at Tokyo Institute of Technology and from 2008, he is an Associate Professor at the same university. He received the Young Engineer Awards both from IEICE and IEEE AP-S Japan chapter in 2001 and 2002 respectively, and Outstanding Paper Award both from SDR Forum and IEICE in 2004 and 2005, respectively and Tutorial Paper Award from IEICE communication Society in 2006. His current research interests are in MIMO propagation measurement, MIMO communication system, MIMO mesh network, and cognitive radio. Assoc. Prof. Sakaguchi is a member of IEICE and IEEE.

Professor Kiyomichi Araki

Prof. Araki (left) was born in 1949. He received the B.S. degree in electrical engineering from Saitama University, in 1971, and the M.S. and Ph.D. degrees in physical electronics both from Tokyo Institute of Technology in 1973 and 1978 respectively. In 1973-1975, and 1978-1985, he was a Research Associate at Tokyo Institute of Technology, and in 1985-1995 he was an Associate Professor at Saitama University. In 1979-1980 and 1993-1994 he was a visiting research scholar at University of Texas, Austin and University of Illinois, Urbana, respectively. Since 1995 he has been a Professor at Tokyo Institute of Technology. His research interests are in information security, coding theory, communication theory, ferrite devices, RF circuit theory, electromagnetic theory, software defined radio, array signal processing, UWB technologies, wireless channel modeling and so on. Prof. Araki is a member of IEEE, IEE of Japan, Information Society of Japan and fellow of IEICE.
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Prof. Araki received IEICE Fellow Award in 2007
Characteristic improvement of Direct Sampling Mixer

Recently, RF circuits have been migrated to deep-submicron CMOS processes. The Direct Sampling Mixer (DSM) is friendly with CMOS technology since it consists of MOS switches and capacitors. Moreover, DSM is one of essential elements in Software Defined Radio (SDR) receiver. Down-conversion and finite-impulse-response (FIR) anti-aliasing filtering are done by charge-sampling along with decimation. Infinite-impulse-response (IIR) channel selection filtering is provided by charge sharing in a discrete charge domain. These operations have reconfigurability, and can reduce power consumption and cost of manufacturing in SDR receiver. In other words, DSM is appropriate for the SDR receiver. However, there are still some problems. We have been improved them from three ways as described in next sections.

Noise Analysis of DSM

The gain of CMOS LNA in the analog front-end is insufficient. The noise components from DSM can not be neglected. Therefore, low noise design of DSM is needed. For the low noise design, noise analysis has been performed. The analysis result of Switched Capacitor Filter (SCF) section in DSM was verified by the simulation results. The result shows that the noise components from SCF section depend on the capacitance value and decimation ratio. In future work, according to the noise analysis, we would show low noise design method for discrete time receiver including DSM.
Analysis of Direct Sampling Mixer as a Multirate System

We analyzed the operation of Direct Sampling Mixer, recognizing it as a linear periodically time-variant multirate system which operates RF sampling, down conversion, filtering, and decimation. To acquire higher-order filtering, we considered the possibility of the polyphase structure of FIR filter and IIR filter in it. We confirmed that we could realize higher-order spatial moving average FIR filtering by improving the structure of capacitances that enabled us to effectively attenuate undesired signals folded due to the decimation.

Non-linear distortion compensation

There are two factors of distortion in DSM. First is the Transconductance Amplifier (TA), where the nonlinearity exists in the amplifier, and which depends on the AM/AM and AM/PM characteristics. Second is the MOS switch, which has nonlinearity due to the Switched Capacitor circuit. The linear and nonlinear characteristics are decided by the charges collected in the capacitor.

Model of DSM

The distortions from MOS switches are included in DSM model. But here, only AM/AM characteristic could be expressed. So, we must construct DSM model which considers not only AM/AM but also AM/PM.
User Scheduling Algorithm for Multi-user MIMO Downlink System

Exploiting of multi-user diversity is already extensively done in wireless SISO communication systems. With user scheduling approaches, average total system capacity will increase with the number of users [1]. The basic idea is to always select the user who owns the best channel condition to transmit. Of course, multi-user diversity can also be exploited in MIMO system, and how to exploit both spatial and multi-user diversity gain has become an interesting topic. With enough users, the simplest receiver similar to zero forcing (multi-user MIMO link) will get more capacity than a single user MIMO link under Eigen mode transmission (EMT) [3]. In this paper, we propose an integrated user scheduling algorithm that combines EMT and ZF receiving. We employ EMT to one selected user who owns the largest first eigenvalue, and other users employ ZF receiving with respect to their channel matrix combined with V matrix of the first selected user (“first user”). We will show that the MIMO channel can be decoupled into parallel channels (sub-channels), and besides the first sub-channel that has been assigned to the first user, all users also can compete independently for each other sub-channel. Finally, we will compare the performance of the proposed algorithm to other approaches.

Simulation result

Below figure shows the simulation results of a 2×2 MIMO downlink system, here both the BS and UEs have two antennas. In the ZF-DP case, we do not care about the antenna number of UE, but only about the total number of receive antennas. In ZF-DP the total number of receive antennas is the same as other approach, which is twice the number of users.

![Fig.1 Capacities of different algorithms (SNR= 0 dB)](image)
Distortion compensating techniques for nonlinear amplifiers in MIMO systems

In communication systems, amplifiers are always used to amplify signals. According to the amplifier’s theory, we can get high power efficiency when the input signal power is near to the amplifier saturation point. However, the amplifier characteristics will become nonlinear and nonlinear distortions will be generated in the output signal. To obtain the high power efficiency and overcome the nonlinear distortions in the amplifier, distortion compensating method is needed when we amplify the signal in the communication system.

In MIMO systems, multi-stream data is transmitted and the input signal in different branch has different input power, frequency and phase. So it is necessary for us to design a new compensation technique for amplifiers in the MIMO system.

High Efficiency Adaptive Modulation Coding

In wireless communications, the presence of reflectors in the environment surrounding a transmitter and receiver create multiple paths which leads to variation in channel status. Adaptive Modulation and Coding (AMC) increase the throughput of wireless network by adjusting the modulation coding scheme (MSC) to channel condition. The most popular modulation being used is M-QAM where M is a power of 2, such as 4, 16, 32, 64, etc. For M=8, the popular modulation scheme is 8-PSK.

Above figure shows the capacity of some modulation schemes in AWGN channel as well as the Shannon limit. There are big gaps between these modulation capacity. It is obvious that with more modulation codes for adaptive modulation scheme, the better performance and the more flexibility we can have.

Our research aims to find more M-QAM schemes where the number of signal points is not a power of 2 to fill the gap between conventional M-QAM, then evaluate these modulations with FEC coding. After that we will consider the effects of power amplifier back-off on these MSC to evaluate some proposed AMC schemes.
Distributed MIMO Networks

Introduction

Project on distributed MIMO networks has been started from 2006 in MIMO group of Araki Sakaguchi laboratory. In a distributed MIMO network, wireless nodes with MIMO antenna will cooperate with each other to improve network spectral efficiency. We are working on the project from theory and algorithm development to implementation and measurements. Two main topics in this project are “Base station cooperation MIMO” and “MIMO mesh network.” Here, we will summarize achievements in the research on MIMO mesh network in 2007.

MIMO Two-way Transmission for Mesh Network

Wireless mesh networks, consisting of mesh routers and clients, have been getting much attention in the fields of sensor networks and wireless plant control systems. Advantages of a mesh network are connectivity, robustness, and wide area coverage owing to multi-hop relay property. Since there are multiple transmission links in a network, multiple access interference occurs, and it degrades throughput performance of the network severely. Multi-channel strategy or Carrier Sense Multiple Access (CSMA) have been commonly introduced to avoid multiple access interference so far.

However, these schemes are not appropriate for high traffic mesh networks, since it is based on FDMA or TDMA based schemes. In this study, we have proposed a new transmission algorithm based on distributed MIMO to achieve high spectrum efficiency even in mesh or multihop networks.

In a considered mesh network, multiple antennas are employed in the mesh nodes to achieve interference cancellation and link multiplexing. The signal flow of proposed MIMO two-way transmission for mesh network is shown in Fig.1. In the proposed MIMO two-way transmission, the interference signals are cancelled by using transmit and receive antenna weights, and forward and backward links are spatially multiplexed. Furthermore, transmit and receive diversity can be achieved by using nonlinear MIMO algorithm such as Dirty Paper Coding (DPC) at the transmitter and Successive Interference Cancellation (SIC) at the receiver.
MIMO Network Coding for Multihop Network

The MIMO two-way transmission algorithm can be extended to MIMO network coding in the case of a multihop network. The signal flow in the MIMO network coding is shown in Fig.2. In the MIMO network coding, the forward and backward streams are combined by using network coding at the relay node, and the combined signal is broadcasted both to the forward and backward links. Since the relay node does not perform the transmit beamforming as in the MIMO two-way transmission, the MIMO network coding does not need a Channel State Information at the Transmitter (CSIT) any more. Moreover, the required number of antennas is smaller than that of MIMO two-way transmission, since the number of interference signals is reduced via network coding.

Furthermore, MIMO network coding can be combined with STBC broadcast to achieve transmit diversity. In the receiver node, virtual MIMO antenna array is formed based on the structure of STBC, and STBC signals from forward and backward links are detected by using normal MIMO algorithm. Thus, the MIMO-STBC network coding realizes high spectral efficiency and high reliability with reasonable system and hardware complexity.

Simulation Results & Hardware Development

The performance of proposed schemes is compared with three conventional schemes: a single antenna multihop network with dual channels, a single antenna multihop network with network coding, a link-by-link MIMO multihop network with dual channel. From Fig.3, it is confirmed that the proposed schemes have extremely higher end-to-end capacity. It is noted that the MIMO two-way transmission with non-linear algorithm has best performance, while the MIMO network coding approaches the MIMO two-way transmission with reasonable system and hardware complexity.

Now, we are developing prototype hardware for the MIMO mesh network. The hardware is operated at 950MHz frequency band in Japan, where the standardization of IEEE802.15.4d for active sensor networks is ongoing.
A 4x4MIMO Fading Simulator (MIMO-FS) was constructed for the development of MIMO products. This MIMO-FS consists of multiple functional boards for 2.4/5.2GHz radio frequency (RF) conversion and baseband digital processing, allows the implementation of various channel models. The architecture of the main system is shown in Figure 1. MIMO-FS can be connected to DUTs (Tx and Rx) instead of antennas to form a 4 x 4 MIMO system. A Channel model can be chosen from GUI for simulating. Moreover, propagation parameters and antenna parameters also can be customized suitable for various simulated environments.

The RF signals from Tx are at first modified by controllable attenuators and converted to IQ baseband signals at downconverter. In the digital circuit, these signals are sampled and multiplied with the channel coefficients generated by fading simulator circuits. The output signals from the digital circuit are up-converted back to RF signals and modified at controllable attenuators before going into Rx. The digital circuit of MIMO-FS consists of fading generator, pre-postprocessors and channel model controllers.

- The channel model controller controls propagation model parameters and RF attenuators. The channel parameters also can be modified from a GUI on PC through LAN and USB interface.
- The fading generator part contains 8 FPGA boards, each can implement the fading generator circuit for 2 channel coefficients with 18 delay tap.

Examples of constellation and spectrum of output signals are shown in Fig. 2. The 40MHz bandwidth QPSK signals were used as input signals while channel model (a) (one path delay) and channel model (b) (6 paths with max delay of 2.51s) were generated inside the MIMO-FS.
Base Station Cooperation between Cellular and Fixed Satellite Service Systems

Base station cooperation (BSC) transmission has been shown to significantly increase average and cell-edge throughputs compared to conventional single-cell transmission. In our work, we extend BSC techniques to cooperatively force full or partial nulls at victim receiver antennas (e.g. Fixed Service Satellite Earth Station or ES) while providing cooperative stream transmission to intended receivers (e.g. Next Generation Cellular user terminals). First, we express null-beamforming (NB) spectrum sharing as a constrained optimization problem where the objective is to maximize capacity or coverage of the intended system and the constraints are the BS transmit powers and intersystem interference levels. We then introduce single-cell null-beamforming (SC-NB), linear BSC-NB, and non-linear BSC-NB algorithms. Finally, we test these algorithms on a 2x2 MIMO, 2-cell, IMT-Advanced physical layer model and show their significant increases in capacity and coverage over a power-control scheme and single-cell null-beamforming schemes.

\[
\begin{bmatrix}
   y_U \\
   y_E
\end{bmatrix} =
\begin{bmatrix}
   H_{UB} & h_{US} \\
   H_{EB} & h_{ES}
\end{bmatrix} \begin{bmatrix}
   x_B \\
   x_S
\end{bmatrix}
\]

where \( y_U \) represents the received UT signals
\( y_E \) contains the received ES signals
\( x_B \) represents the transmitted signals at the BSs
\( x_S \) is the transmitted signal at the satellite
\( H_{UB} \) are the MIMO channels between the BSs and UTs
\( h_{US} \) is the SIMO channel between the satellite and the UTs
\( H_{EB} \) are the MISO channels between the BSs and the ESs
\( h_{ES} \) are the SISO channels between the satellite and the ESs

User Spectral Efficiency Versus Location (2x2 MIMO)

BS 1 (BS2) is located 0.7 kms (2.1 kms)
ES is located 1.05 kms (from left side)

Allowed interference limit at the ES is -130.6 dBm
Contributions

Suzuki-Fukawa Laboratory

Transactions and Letters


International Conferences


Domestic Conferences


Takada Laboratory

Transactions and Letters


International Conferences


References


Domestic Conferences


Araki Sakaguchi Laboratory

Transactions and Letters


International Conferences


Domestic Conferences


