IMPULSE HIT

Content

Vision

IMPULSE HIT is a yearly newsletter to share upcoming technologies in various fields of Electronics & Communication Engineering. This issue of newsletter introduced the advanced research topics in the field of antenna system for interference rejection, speech processing as well as cloud networks. These brief technical articles will influence many students and faculty members to extend research works in these fields in near future.

Team Members

Switched Beam Smart Antenna System

Hema Anand, 4thYear, ECE

The switched beam system employs an antenna array which generates fixed overlapping beams that cover the certain angular area. The switched beam system is employed in the base station, when an incoming signal is detected by the base station, the system selects the right beam which is best aligned to the signal-of-interest direction and then switches to that beam to communicate with the user. The selection of right beam is usually based on the maximum received power for that user. A typical cellular area is divided into three sectors with 120° angular width. Each sector served by the six overlapping fixed directional narrow beams as shown in Fig. 1. The spatially separated directional beams can significantly reduce the co-channel interferences which lead to increase the frequency reuse as well as the capacity of the cellular system. The uplink and downlink communication can be served by the same beam.

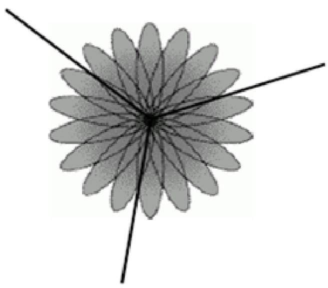


Fig. 1. Coverage pattern of switched beam antenna system.

The radiation performance of the switched beam antenna system is shown in Fig. 1. A typical switched beam system for a base station would consist of multiple arrays with each array covering a certain sector in the cell. It consists of a phase shifting network, associated with the antenna array as shown in Fig. 2, forms multiple directive beams in different directions. The selection of the right beam which is best aligned to the signal of interest direction is made by the control logic. The control logic is driven by an algorithm, scans all the beams and selects the suitable beam which receives the strongest signal. The detector makes the measurement, which beam receives the strongest signal. Based upon this data of the detector the control logic selects the right beam. Finally switch to the right beam is made by the RF switch.

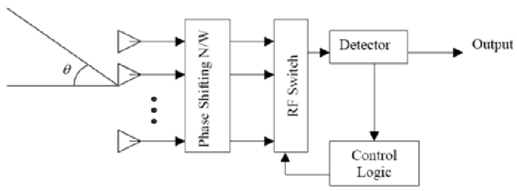


Fig. 2. Basic block diagram of a typical switched beam system.

The working mechanism of switched beam system is simpler than the adaptive array but the system is not suitable for high interference areas. Let us consider a user is served by a directional beam. When another user is at the direction of the broad null then there would be no interference from the second user. But if the second user enters into the same area of the beam as the first user, then that user could cause interference to the first user. Therefore switched beam systems are best suited for a small or zero-interference environment. Although this is a disadvantage of the switched beam system but it provides a significant enhancement in capacity of the wireless system, considerable interference rejection when the desired user is at the centre of the beam. Also, it is less expensive and can be easily implemented in older systems.

Challenges of Audio and Speech Processing

Ritambhara, 3rd Year, ECE

The objective of our research is to design speech recognition systems that are capable of analyzing and processing auditory scenes.

Although humans can naturally analyze auditory scenes into meaningful events, and consciously or passively respond to them, computer audition remains a challenging problem and once it succeeds, it will represent a fundamental advance in Audio Signal Processing and Intelligent Systems. Successful computer audition systems will also improve our ability to access and manipulate audio data, enabling new multimedia applications and interactions for experts and novices alike. Auditory scenes are difficult to parse because source signals composing them overlap and interfere with each other at the same time and frequency. For example, when one is walking on a street, the auditory signal that he/she receives at any time leaves, people chatting, and a car passing by.

Furthermore, the ways that these sources mix together and the source signals themselves are constantly changing. Imagine recognizing objects in a visual scene where every object is half-transparent (sound sources overlap), objects change their appearance (sound sources change over time) and even disappear or reappear unexpectedly (sound sources become silent and active). A successful speech recognition system must be able to focus source signals in composite auditory scene in spite of the other interfering sources. This challenge raises two fundamental questions: 1) How do we discover and separate a sound event from an audio mixture? 2) What sound events belong to one source versus other sources? To answer these questions, I am interested to do research at three levels:

- Focusing on source sound by using the speech perception property of human ear. (e.g. identifying footsteps and streaming them together)

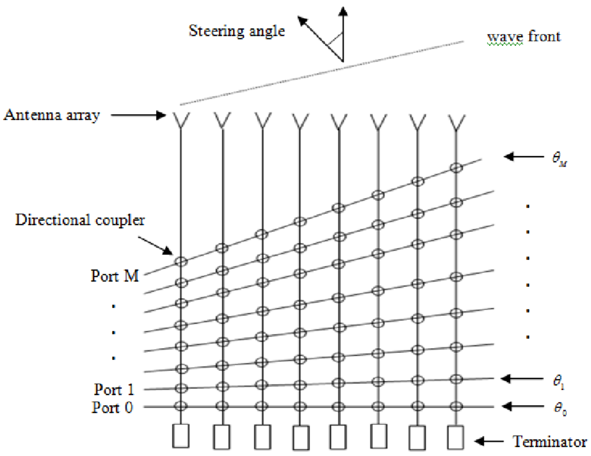
- Designing the new front end features by employing the human auditory property. (e.g. extracting vocals from music, removing noise to enhance speech)

- Leveraging external information (e.g. a musical score or speech transcript) to improve separation and manipulation of sound objects and sources.

Blass Matrix

Surajit Mukherjee, Assistant Professor, ECE

Blass matrix is a phase shifting network. The Blass matrix consists of directional couplers and transmission lines to provide the necessary phase shift between the output ports of the matrix. A Blass Matrix, which fed the 8 elements array to produce directive beams in different directions, has been presented in the figure below. Each node represents the directional coupler which is used to cross-connect the transmission lines. When the signal is fed to the input ports, the specific phase differences between the output ports of the matrix will enable the beams direct to the different steering angle, which are generated by the antenna array. Though the configuration of Blass matrix is simple but the performance of the matrix is not satisfactory due to the losses attributed to the resistive terminations.



Beam forming network implemented by Blass matrix array.

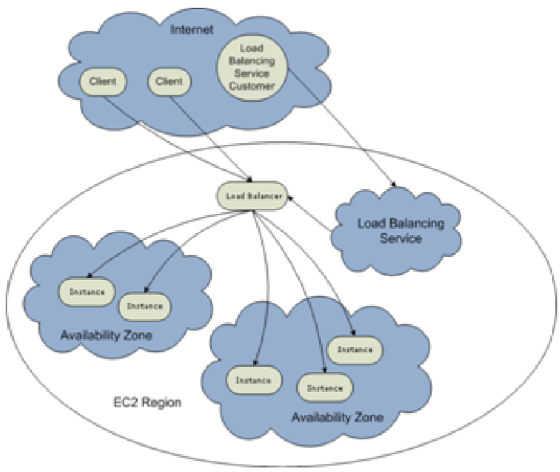
Load Balancing in Cloud Networks

Malay Kumar Pandit, Professor, ECE

In current cloud networks, load balancing is essential for optimum utilization of the servers. It is achieved by optimum allocation of jobs among the pool of servers. At a point of time, some of the servers are found busy while some are sparingly loaded. Balancing is enforced by several means. One effective method is by statistically segmenting the load. Another technique is to differentiate and distribute traffic according to the traffic pattern. It is a kind of software that routes the different traffic flows to appropriate servers.

Principal performance parameters considered are packet loss rate (PLR), end-to-end delay, bandwidth and timing jitter. The packets are lost due to missed deadlines, limited buffer size and unacceptable rise in waiting time. Keeping end-to-end delay within limit is important for maintaining integrity of the incoming traffic. Throughput is important for complying with the SLA (service level agreement).

Machine learning techniques are on the rise for efficient load balancing. The traffic patterns are learnt and appropriate routing and processing are carried out. The traffic may be Poisson or self-similar fractal type. Intelligent Load balancing uses the concept of auto-scaling of resources. This translates to offering resources based on demands. The configuration time is kept of the order of 50 ms, the resources are offered from a common pool. The auto-scaling is carried out depending on the targeted CPU utilization of a group of virtual machine instances. The policy algorithm checks the average CPU utilization of the instance group and decides if it needs to be scaled. If the average value for the cores in the group exceeds the target utilization, it will add more virtual machines to the group. The cool down period is the number of seconds the auto-scalar must wait after a virtual machine starts before it starts



Load balancing in a cloud network.

Beam Forming Antenna System using Butler Matrix Array

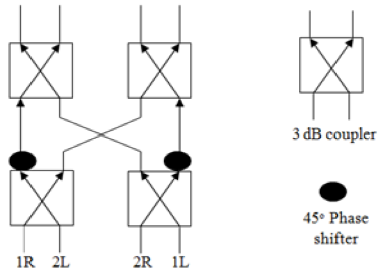
Wriddhi Bhowmik, Assistant Professor, ECE

A $N \times N$ Butler matrix consists of N-element array can produce N directive beams in different directions ($N = 2^n$). The 3 dB coupler and 45° phase shifter are the basic building block of the Butler matrix array. The system requires $2^{n-1}\log_2 2^n$ and $\left(\frac{2^n}{2}\right)\left[\{\log_2 2^n\}-1\right]$ numbers of 3dB coupler and 45° phase shifter respectively. The crossover or 0 dB coupler is used in the path crossing of two transmission lines for the single layer implementation of Butler matrix array. When the input ports of the matrix is excited by a signal, different inter-element phase shifts are obtained between the output ports of the matrix. These phase differences direct the beams in different directions. The antenna elements are connected to the output ports of the matrix, which are used to generate the directive beams. The beams are pointing in direction θ governed by the following equation

$$\theta = \cos^{-1}\left[\frac{\pm(2n-1)\lambda_0}{8d}\right] \qquad (1)$$

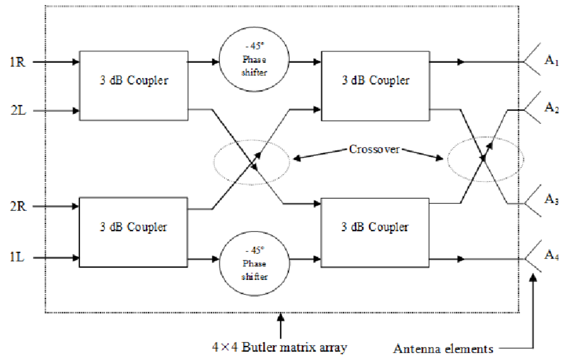
Where n = 1,2... etc, λ_0 is free space wavelength and d is the physical distance between the centers of antenna elements.

Fig. 1(a) and (b) respectively represent the schematic of 4 × 4 Butler Matrix array and a basic block diagram of a 4 × 4 Butler Matrix fed antenna array system. The 4 × 4 Butler Matrix fed antenna array system presented in Fig. 1 (b) can be referred as the multiple beam forming antenna system (MBFAS). As described earlier, the 3dB coupler, 45° phase shifter and the crossover or 0 dB coupler are considered as the basic building blocks of the Butler Matrix array. A 4×4 Butler Matrix consists of four 3 dB branchline couplers and two 45° phase shifters, for the planar implementation of the system two crossovers are required at the crossing of the transmission lines.



(a)

Fig. 1 (a) Schematic of 4×4 Butler matrix array.



(b)

Fig. 1 (b) Basic block diagram of a 4 × 4 beam forming antenna system.

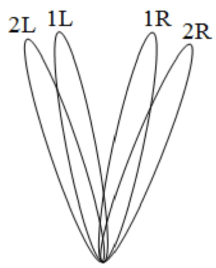


Fig 2. Proposed radiation pattern of 4 × 4 Butler Matrix fed antenna array system.

The output ports of the Butler matrix array feed the antenna elements A_1, A_2, A_3 and A_4 as shown in Fig. 1 (b) to create the directive beams in different directions. The input ports of the system are named as 1R, 2L, 2R and 1L based on the specific direction of the major beams (R: Right and L: Left). The 1R port excitation leads the system to generate the major directive beam in the range of 0^o to 30^o. Similarly remaining major beams directions should be in the range of -30^o to -60^o, 30^o to 60^o and 0^o to -30^o for 2L, 2R and 1L port excitation respectively as proposed in Fig. 2. The pointing major beam directions for 4×4 beam forming antenna system can be calculated from the Eq. (1).

Motivation for Multiple Camera Audio-visual Speech Recognition Research

Aarti Kumari & Aarti Yadav, 2nd Year, ECE

The application of speech recognition for the development of robotics and computer vision is always the first priority of research. The notable progresses of AVSR have been reported in the recent years. However, these systems were hardly encountered with real-time visual domain issues. The primary concern is to track the motion of visual articulator properly during natural movement of body parts. Most of the existing VSR systems have been developed by assuming that speakers keep their head in steady position. In the real time scenario, this is very difficult to keep the head in unmoved condition. The other real-time visual domain constraints are light condition, poor resolution, view angle, head position, occlusions etc. Thus, there is a need to develop a more flexible AVSR system where speakers would have more liberty to move their body parts naturally. It is not possible to have all movement and pose related information with a single camera. Multiple camera protocol is well efficient to capture speaker's natural movement-related information. Multiple camera protocol is a real solution to the visual domain problems where each camera can provide complementary information to another one.

The main burden to the researcher is the availability of the standard multiple camera database that addressed the visual domain issues. It is required to train our system in such a way that it could face visual domain issue. So it is required to develop a standard multiple camera AVSR database in controlled and uncontrolled environment.

However, another challenge of multi-cam AVSR protocol is the camera-fusion approach. Most of the reported techniques have adopted for HMM-fusion approach. The major disadvantage of this technique is the computational cost which could be a major real-time factor. The more practical approach would be to discard the mutual information shared by the each camera. Thus, a hypothetical test between each pair of camera could minimize the computational cost.