4G CHALLENGES

While migrating from 3G to 4G, certain challenges have to be faced.

Multimode user terminal: Multimode user terminal is a device operating in different modes supporting a large type of 4G services and wireless networks by reconfiguring themselves to adapt to different wireless networks. They encounter several design problems like limitations in the device cost, size, backward compatibility to systems and power consumption.

Wireless network discovery: Availing 4G services need the multimode user terminal to find and select the required wireless network. Discovery of 4G systems will be much more challenging than 3G due to the heterogeneity of the networks and their access protocols.

Wireless network selection: 4G will offer the users a option to select a wireless network providing optimized performance and high Qi's for a specific place, time and desired service (communication, multimedia). But the parameters that define high Qi's and optimized performance at

Specific instant must be clearly defined to form the network selection procedure efficient and transparent to the end user.

Terminal mobility: Terminal mobility is an important characteristic to satisfy the "Anytime Anywhere" promise of 4G. It permits the mobile users to move across the geographic boundaries of wireless networks. Two important problems in terminal mobility are location and hand off management. Location management includes tracking the location of the mobile users and maintaining data like the authentication information, Qi's capabilities, and the original and the current cell location. Handoff management is maintaining the

Continuing communication when the terminal roams.

Network infrastructure and Qi’s support: Unlike previous generation networks such as 2G and 3G, 4G is an integration of IP and non-IP based system. Before 4G, Qi’s designs were made with a specific wireless system in mind. But in 4G systems Qi’s designs should consider the integration of various wireless networks to ensure QoS for the end-to- end services.

Security: Most of the security schemes and the encryption/decryption protocols of the present generation networks were designed only for specific services. They appear to be very inflexible to be used across the heterogeneous architecture of 4G that desires dynamically adaptive, reconfigurable and light-weight security mechanism.

Fault tolerance: Wireless networks resemble a tree-like topology. Any failure in one of the levels will affect all the network elements at the levels below. This problem may become more complex because of the multiple tree topologies. Adequate research work is needed to devise a method for fault tolerance in wireless networks.

APPLICATIONS OF 4G

- **Virtual presence:** 4G will provide user services at all times, even if the user is off-site.

- **Virtual navigation:** 4G will provide users with virtual navigation through which a user can access a database of streets, buildings, etc., of a large city. This requires High-speed transmission.

- **Tele-medicine:** 4G will support the remote health monitoring of patients via video conference assistance for a doctor anytime and anywhere.

- **Tele-geo-processing applications:** 4G will combine geographical

Information systems (GIS) and global positioning systems (GPS) in which a user will get location querying.

- **Education:** 4G will provide a good opportunity to people anywhere in the world to continue their education on-line in a cost-effective manner.

- **Multimode Software Application**

Multimode software is software that allows the user device to adapt itself to various wireless interfaces networks in order to provide constant net access with high data (packet based) rate.

All the networks will be compatible once the switch is completed, eliminating roaming and areas where only one type of phone is supported. Once the voice and data networks are superposed there will suddenly be millions of new devices on the network cloud. This will require either reconstruction of the address space for the entire Internet or using different address spaces for the existing wireless networks. The multimode device architecture may improve call completion and expand effective coverage area.

- **Support for Multiple and Efficient Applications and Services-** 4G provides support for unicast, multicast and broadcast services and the applications that rely on them. Prompt enforcement of Service Level Agreements (SLA) along with privacy and other security features.

- **Quality of Service -**Consistent application of admission control and scheduling algorithms regardless of underlying infrastructure and operator diversity leads to an increased quality of service (Qos) to the users.

- **Network Detection Selection**: A mobile terminal that features multiple radio technologies or possibly uses software defined radios if economical, Allows participation in multiple networks simultaneously, thereby connecting to the best network with the most appropriate service parameters (cost, Qi's and capacity among others) for the application. This requires establishing a uniform process for defining eligibility of a terminal to attach to a network and to determine the validity of link layer configuration.

- **Handover and Service Continuity:** A base station that features intra- and inter-technology handovers, assuring service continuity with zero or minimal interruption, without a noticeable loss in service quality. Support for this function requires continuous transparent maintenance of active service instances and inclusion of various access technologies, from Wi-Fi to OFDMA.

- **Crisis Management Application**: In the event of natural disasters where the entire communications infrastructure is in disarray, restoring communications quickly is essential. With wideband wireless mobile communications, limited and even total communication capability (including Internet and video services) could be set up within hours instead of days or even weeks required at present for restoration of wire line communications.

ADVANTAGES OF 4G:-

1. Support for interactive multimedia services like teleconferencing
2. Wider bandwidths and higher bitrates.
3. Global mobility and service portability.
4. Scalability of mobile network.
5. Entirely Packet-Switched networks.
6. Higher band widths to provide multimedia services at lower cost(up to 100Mbps).
7. Tight network security

4G TECHNOLOGIES

MULTICARRIER MODULATION

Multi-carrier modulation (MCM) is a method of transmitting data by splitting it into several components, and sending each of these components over separate carrier signals. The individual carriers have narrow band width , but the composite signal can have broad bandwidth.

The advantages of MCM include relative immunity to fading caused by transmission over more than one path at a time (multipath fading), less susceptibility than single-carrier systems to interference caused by impulse noise, and enhanced immunity to inter-symbol interference. Limitations include difficulty in synchronizing the carriers under marginal conditions, and a relatively strict requirement that amplification be linear.

Multicarrier modulation (MCM) is a derivative of frequency-division multiplexing. Forms of multicarrier systems are currently used in DSL modems and digital audio/video broadcast (DAB/DVB). MCM is a baseband process that uses parallel equal bandwidth sub channels to transmit information and is normally implemented with fast Fourier transform (FFT) techniques. MCM’s advantages are better performance in the inter symbol-interference environment and avoidance of single-frequency interferers.

However, MCM increases the peak-to-average ratio of the signal, and to overcome inter symbol interference or guard band must be added to the data. The difference, D, of the peak-to-average ratio between MCM and a single carrier system is a function of the number of subcarriers, N, as:

D (dB) = 10 logN

Any increase in the peak-to-average ratio of a signal requires an increase in linearity of the system to reduce distortion. Linearization techniques can be used, but they increase the cost of the system.

If L_b is the original length of block and the channel's response is of length L_c, the cyclically extended symbol has a new length $L_b + L_c – 1$. The new symbol of length $L_b + L_c – 1$ sampling periods has no inter symbol interference. The cost is an increase in energy and encoded bits are added to the data.

Two different types of MCM multicarrier code division multiple access (MC- CDMA) and orthogonal frequency-division multiplexing (OFDM) using time- division multiple access (TDMA). MC-CDMA is actually OFDM with a CDMA overlay.

Similar to single-carrier CDMA systems, the users are multiplexed with orthogonal codes to distinguish users in MC-CDMA. However, in MC-CDMA, each user can be allocated several codes, where the data is spread in time or frequency. Either way, multiple users simultaneously access the system.

In OFDM with TDMA, the users are assigned time slots to transmit and receive data. Typically MC-CDMA uses quadrature phase shift keying (QPSK) for modulation, while OFDM with TDMA could use more high-level modulations, such as multilevel quadrature amplitude modulation (M-QAM. In OFDM the subcarrier pulse shape is a square wave. The task of pulse forming and modulation is performed by a simple inverse FFT (IFFT) which can be implemented very efficiently. To decode the transmission, a receiver needs only to implement FFT.

The OFDM divides a broadband channel into many parallel sub channels. The OFDM receiver senses the channel and corrects distortion on each sub channel before the transmitted data can be extracted. In OFDM, each of the frequencies is an integer multiple of a fundamental frequency. This ensures that even though sub channels overlap, they do not interfere with each other.

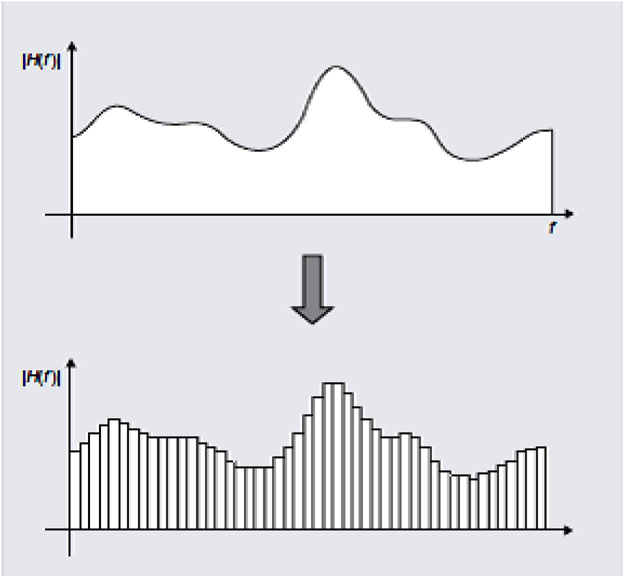

Fig. 5.3: A broadband channel divided into many parallel narrowband channels.

[Source: Text book- Wireless Communications and networking First Edition, Elsevier 2007 by Vijay Gag]

THE 4G TRANSCEIVER:

The structure of a 4G transceiver is similar to any other wideband wireless transceiver. A multicarrier modulated signal appears to the RF/IF section ofthe transceiver as a broadband high PAVR signal. Base stations and mobiles aredistinguished in that base stations transmit and receive/ decode more than one mobile, while a mobile is for a single user. A mobile may be a cell phone, a computer, or other personal communication device. The line between RF and baseband will be closer for a 4G system. Data will be converted from analog to digital or vice versa at high data rates to increase the flexibility of the system. Also, typical RF components such as power amplifiers and antennas will require sophisticated signal processing techniques to create the capabilities needed for broadband high data rate signals.

The following figure shows a typical RF/IF section for a transceiver. In the transmit path in phase and quadrature (I&Q) signals are up converted to an IF, and then converted to RF and amplified for transmission. In the receive path the data is taken from the antenna at RF, filtered, amplified, and down converted for baseband processing. The transceiver provides power control, timing and synchronization, and frequency information. When multicarrier modulation is used, frequency information is crucial. If the data is not synchronized properly the transceiver will not be able to decode it.

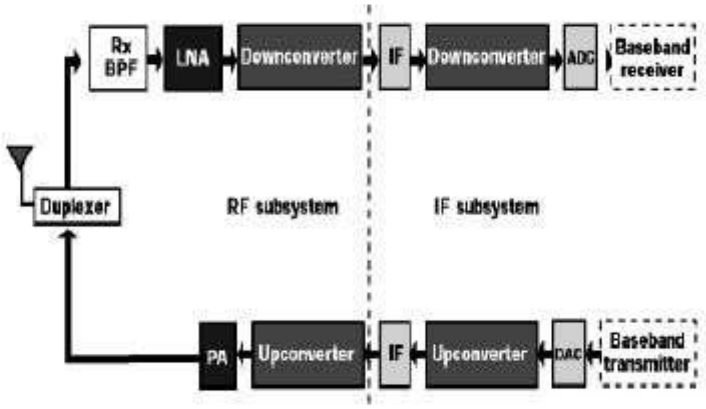

Fig.5.4: RF/IF Block Diagram

[Source: Text book- Wireless Communications and networking ,First Edition, Elsevier 2007 by Vijay Gag]

RECEIVER SECTION:

4G will require an improved receiver section, compared to 3G, to achieve the desired performance in data rates and reliability of communication. The minimum required SNR for reliable communication:

$SNR=2^{C/BW}$

Where C is the channel capacity (which is the data rate), and BW is the bandwidth for 3G, using the 2-Mbps data rate in a 5-MHz bandwidth, the SNR is only 1.2 dB. In 4G, approximately 12-dB SNR is required for a 20-Mbps data rate in a 5-MHz bandwidth. This shows that for the increased data rates of 4G, the transceiver system must perform significantly better than 3G. The receiver front end provides a signal path from the antenna to the baseband processor. It consists of a band pass filter, a low-noise amplifier (LNA), and a down converter.

De-pending on the type of receiver there could be two down conversions (as ina super-heterodyne receiver), where one down conversion converts the signal to an IF. The signal is then filtered and then down converted to or near baseband to be sampled. The other configuration has one down conversion, as in a homodyne (zero IF or ZIF) receiver, where the data is converted directly to base band. The challenge in the receiver design is to achieve the required sensitivity, intermodulation, and spurious rejection, while operating at low power.

APPLICATIONS OF MCM

- In analog military communications
- Digital audio and video broadcast services
- Digital television ,
- Obtaining high data speeds in asymmetric digital subscriber line (ADSL) systems.
- MCM is also used in wireless local area networks (WLANs).
- Fixed wireless broadband services;
- Mobile wireless broadband communications
- Multiband OFDM for ultra wideband (UWB) communications;

Introduction to Broadband Wireless Access

The focus of meeting the growing need for broadband access has shifted onto cover rural and remote areas with relatively lower user density and no network infrastructure. Broadband Wireless Access (BWA) is now considered a promising alternative due to the availability of low cost commodity wireless hardware, newly freed spectrum and the progress in communication technology.

Wireless Optical Broadband Access Networks (WOBAN) is a popular BWA architecture. It has a wired optical backhaul network at back end and a wireless mesh network at front end. The wireless part provides access to end users and the wired part carries the aggregated traffic from the wireless part. Such an architecture is shown in Figure 1. The optical backhaul network consists of a Passive Optical Network (PON), an Optical Line Terminal (OLT) in the Central Office (CO) that is connected to multiple Optical Network Units (ONUs) via optical fiber. At the front end, end users access the network through the wireless mesh that has stationary gateways. The user traffic travels over multi-hop through the gateway and reaches ONUs.

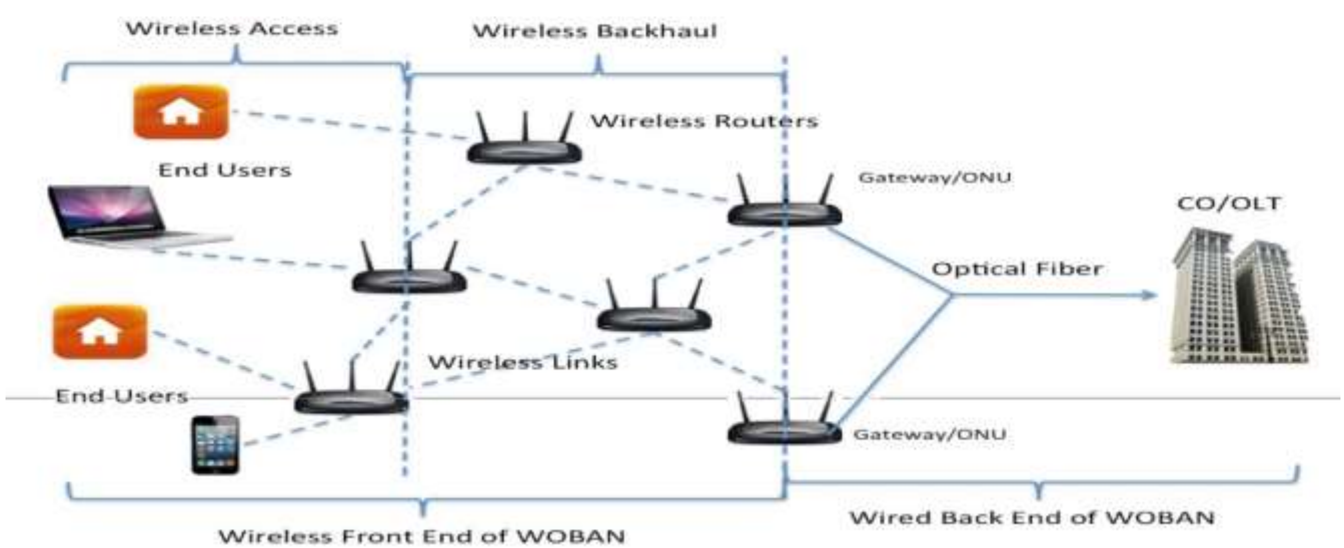

Figure5.16: WOBAN Architecture.

[Source: Text book- Wireless Communications and networking , First Edition,Elsevier 2007 b

vijay Garg]

There are multiple research efforts aimed at improving WOBAN such as management system and energy efficiency. Therefore the WOBAN architecture serves as a base for later discussion from the above aspects. Other researchers refer to WOBAN as FiWi BAN (Fiber Wireless Broadband Access Networks). To maintain the consistency between this survey and the references, we use the same terminology as the source. Therefore the term FiWi BAN may appear when discussing relevant works.

General BWA networks design involves several topics. The fundamental one, firstly, is to improve the capacity of the access network. Various works have been proposed including optimal relay station placement, bandwidth allocation scheme and direct communication mode between subscriber stations. Other relevant works focus on management systems for large-scale BWA networks and the performance comparison of multiplex techniques for 4G.

Another crucial concern when considering deploying BWA networks is energy and cost efficiency. Reducing the energy consumption not only saves electric billing but also helps to reduce carbon dioxide emission and is environment-friendly. Lots of research efforts have been put into this issue such as the study of energy efficiency from an architectural level, a mixed capacity access proposal, the study for reducing energy in long reach access and a green WOBAN design. Other important topics on BWA include the enhancement to survivability and specific application scenarios that can benefit from BWA networks.

The rest of the paper is organized as follows. Section 2 covers the enhancements proposed to improve the capacity, utilization and management of BWA networks. Section 3 is dedicated to a specific topic concerning the energy and cost efficiency of future deployments of BWA networks. A few application scenarios are discussed in section 4 and finally section 5 concludes this paper.

Enhancements Proposals

There are ongoing research efforts dedicated to improve BWA networks from various aspects. These topics includes the management system for large-scale networks, the optimal relay station placement scheme aimed at improving the network capacity and bandwidth utilization, the bidirectional bandwidth allocation approach for TCP in BWA networks, the adaptive direct communication mode between base stations and subscriber stations and the performance comparison between multiplex techniques. These research efforts are discussed in the following subsections respectively.

Management for Large Scale BWA

Expanding the coverage of the Internet to the next billion people would naturally lead to BWA networks of very large scale. Given the wide range of parameters and environmental phenomena that affect network operation, like many general wireless networks, BWA networks are inherently difficult and complex to manage. BWA networks are also be quite diverse, so network management platforms should be flexible to suit varying deployment requirements. To manage large-scale BWA networks, dedicated management approach maybe infeasible due to the huge additional cost and deployment overhead required. Therefore, simplifying network management is crucial to BWA. It is also a key requirement to simplify the specification of management goals for an effective distributed management system. Proposes the Stix platform to manage large-scale BWA networks. The management of BWA networks includes a wide range of activities such as fault, configuration, accounting, performance and security. Stix focuses on performance, fault and configuration management. It eases the management of community and ISP deployments while keeping the infrastructure scalable and flexible. Based on the notions of goal-oriented and in-network management, Stix allows administrators to graphically specify network management activities as workflows. Administrators are deployed at a distributed set of agents within the network that cooperate in executing those workflows and logging management information.

Figure shows the architecture of a Stix agent. The communication manager is connected to other Stix agents. It is responsible for listening on incoming messages and forwarding them as workflows to the correct internal part. A workflow manager receives new workflows from the communication manager and determines the relevancy to local devices under management. If so, it stores the workflow, transforms them into proper forms and pass them down to the workflow engine. The workflow engine registers and schedules the workflow execution. The log overlay records local or neighboring log messages. The storage manager provides interfaces for appropriate database persistent storage. The device manager communicates with locally managed devices.

Stix is implemented on embedded boards and has a low memory footprint. The implementation generates operating topology and traffic data, showing that Stix can reduce the management traffic significantly, in comparison to common centralized management approach. Case studies are also used to demonstrate the ease of deploying the Stix platform for network reconfiguration and performance management. The author therefore point out the potential of Stix to be self-managing.

Relay Station Placement Bandwidth and channel interference limit the kind of approaches that aim at maximizing resource utilization.

Alternatively, cooperative relaying has been seen as one of the effective solutions to satisfy the stringent requirement of capacity enhancement for BWA. Deploying Relay Stations (RSs) can improve the quality of wireless channel for two reasons. First, the one long distance low rate link is replaced with multiple short distance high rate links. Second, obstacles between Base Station (BS) and Subscriber Station (SS) that affect channel quality can be effectively circumvented using RSs. As a result, placing RSs

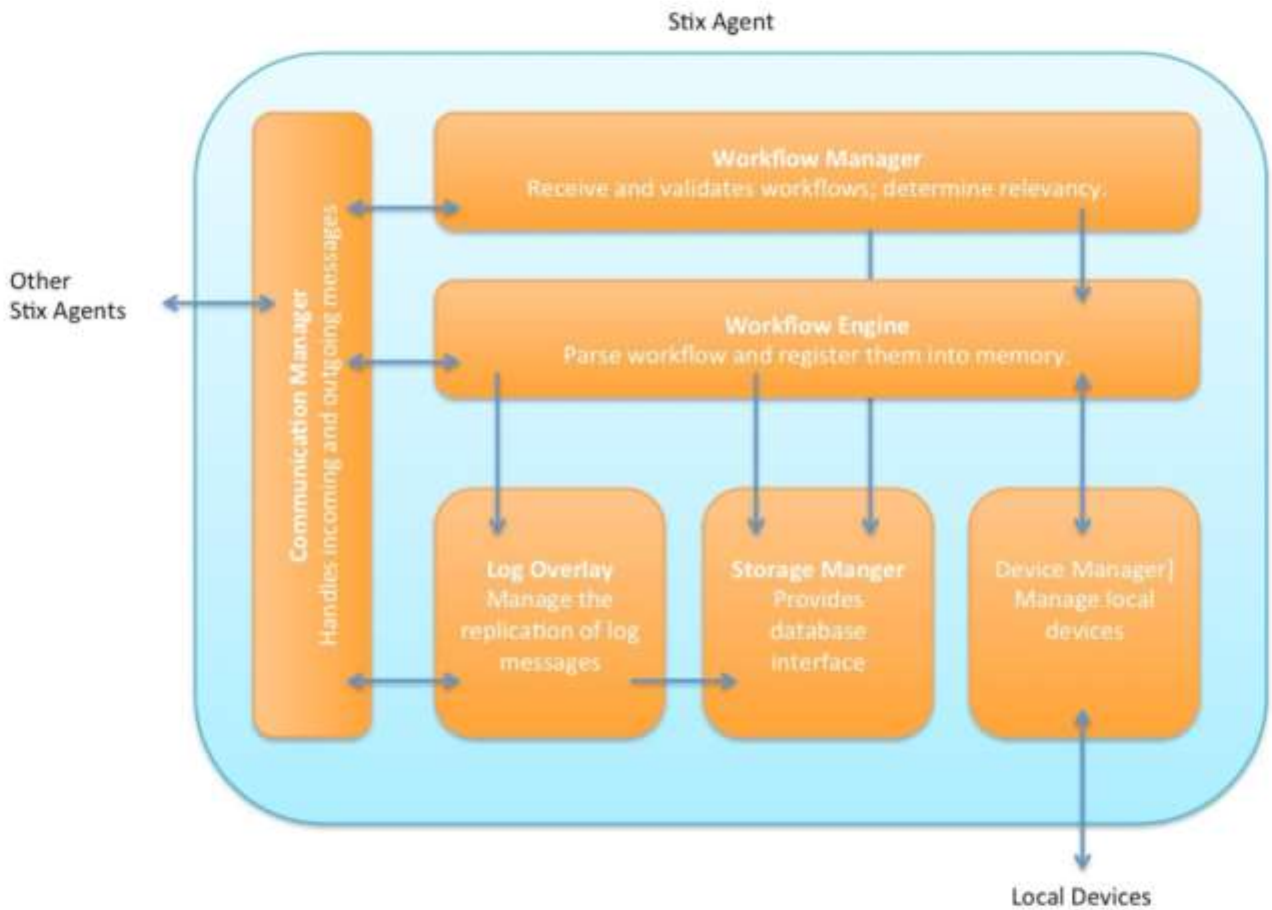

Figure .5.17: Stix Agent Architecture.

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

The RS placement problem and developed an optimization framework to maximize the capacity as well as to meet the minimal traffic demand by each SS. First, a new RS location planning design paradigm is presented. It incorporates the cooperative relaying into the network design. Second, to solve problem of the RS placement for maximum capacity, the authors develop an optimization framework. In comparison to traditional approaches that decouple the RS placement and bandwidth allocation into two separate phases, this framework jointly considers these two phases and output the optimal location placements along with the allocated bandwidth. The authors formulate such problem into MILP and seek solution through CPLEX. In addition, a heuristic algorithm is proposed to avoid the exponential solving time for the problem and produce the solution in polynomial time. Such timely solution can cope with growing traffic responsively.

A series of case studies and the numeric results from the simulation lead to two conclusions. First, cooperative relaying outperforms non- cooperative relaying in capacity given the same number of RSs. Second, the proposed heuristic algorithm solves the MILP problem much faster than the standard CPLEX with only a slight degradation in capacity performance. The cooperative relay placement strategy combining the heuristic algorithm provides an important guideline for future BWA network deployment and capacity planning.

Bandwidth Allocation for TCP

In IEEE 802.16 networks, ACKs are transmitted uplink over a unidirectional connection that is different from the downlink TCP flow. This behavior increases the round trip delay of the TCP flows because additional bandwidth has to be allocated for this ACK connection. The throughput also suffers accordingly. Regarding such behavior, a bidirectional bandwidth-allocation mechanism is proposed in that couples the uplink and the downlink bandwidth allocation and thus increases the throughput of the downlink TCP flow and enhances the efficiency of uplink bandwidth allocation for the TCP ACK.

Firstly, an analytical model is presented to investigate the impact of the uplink bandwidth allocation delay on the downlink TCP performance. The simulation studies show that the piggyback method is not always available due to the burstiness of TCP. The authors conduct further studies to investigate the conditions under which the piggyback method will be available. Results show that when the bandwidth request delay is removed or reduced, the TCP throughput will be increased by up to 30% due to the availability of the piggyback mechanism.

After confirming the degrading effect, the authors propose the bidirectional bandwidth allocation scheme to address such impact. The scheme combines proactive bandwidth allocation with piggyback request to mitigate the drawbacks due to the proactive allocation. Figure shows how the allocation scheme works. When serving a Downlink (DL) frame, the BS scheduler checks the Uplink (UL) queue to see if it is empty. If so, the scheduler enqueues a new bandwidth request for the SS. This is the proactive allocation approach. On the other hand, upon a UL frame, the scheduler checks if the packets are still in the DL data queue. If the queue is not empty, then the SS will request for bandwidth in a piggyback manner.

The combined scheme reduces the unnecessary bandwidth request delay and the signaling overhead because the needed bandwidth under such scheme is proactively allocated. According to simulation results, such allocation scheme can improve TCP throughput up to 40%. Moreover, the scheme is simple and easy to implement in the base station without causing modifications to the subscriber station or change to cross-layer signaling mechanisms.

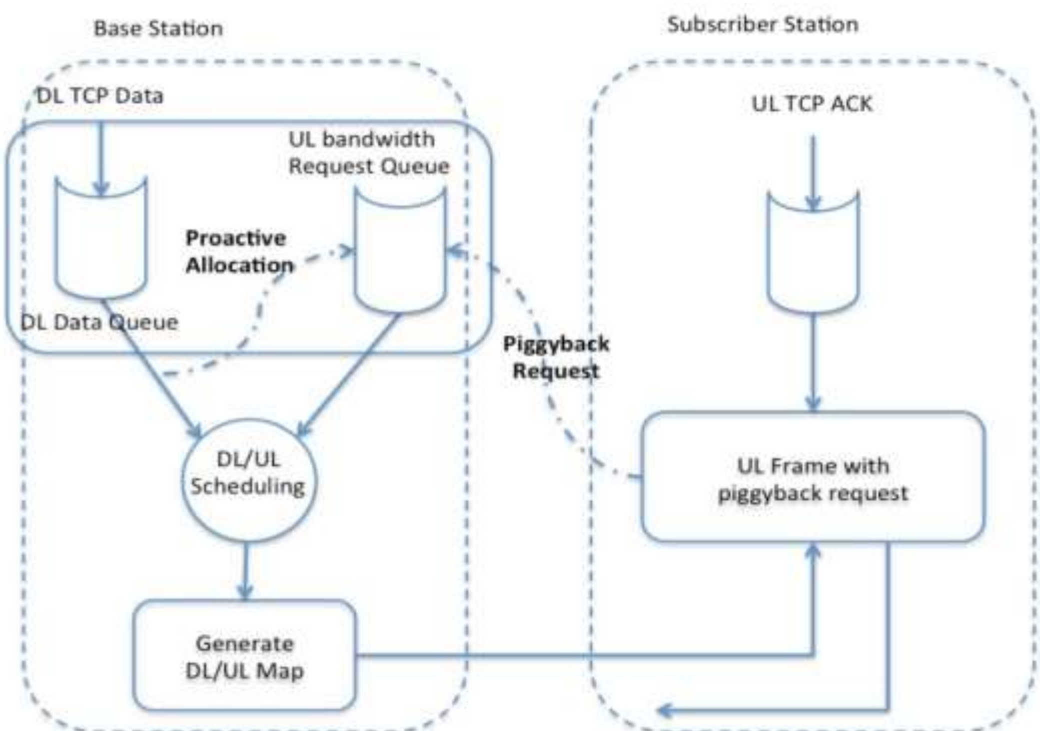

Figure .5.18: Schematic Diagram of the bidirectional bandwidth allocation

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

Adaptive P2P approach for Subscriber Station

In IEEE 802.16 networks, the Point-to-Multipoint (PMP) mode is a well-adopted transmission type. In PMP mode, the Base Station (BS) serves as the centralized coordinator controlling and forwarding packets for the Subscriber Stations (SS). However, when two SSs intend to exchange packets, the BS is still required to reroute the packets. Obviously the communication bandwidth is wasted because of the rerouting. To address this problem, proposes the Adaptive Point-to-point Communication (APC) approach to achieve direct communication between SSs within the PMP mode in IEEE 802.16 networks. Although under their proposal, the BS is still needed to coordinate and arrange time intervals for the SSs to transmit, the bandwidth consuming packets rerouting part is avoided. Given the additional APC mode, the channels can be utilized more adaptively by switching between the PMP and the APC mode.

Figure depicts the switching process for packets transmission in APC. The switching between APC and the conventional indirect communication mode is dominated by certain constraint. Such constraint is calculated based on the channel quality between BS and SSs. Therefore, the APC mode always chooses the most efficient transmission for intra-cell traffic. As a result, the network throughput is expected to be increased using APC.

To confirm such expectation in performance improvement, the author first conduct theoretical calculations to derive the maximum number of MAC SDUs that can be transmitted and received within a frame. Then simulations are performed to validate such numeric model. In order to show the improvement after adding the APC mode, the performance analysis is conducted in a comparative manner. Conventional IEEE 802.16 and the proposed APC mode are compared in terms of saturation throughput in aspects of modulation mode, MAC SDU size and intra-cell traffic flow. The evaluation results show that by adding the APC mode, there is a significant increase in network throughput with the slight increase in network traffic overhead due to the additional coordination needed in BS.

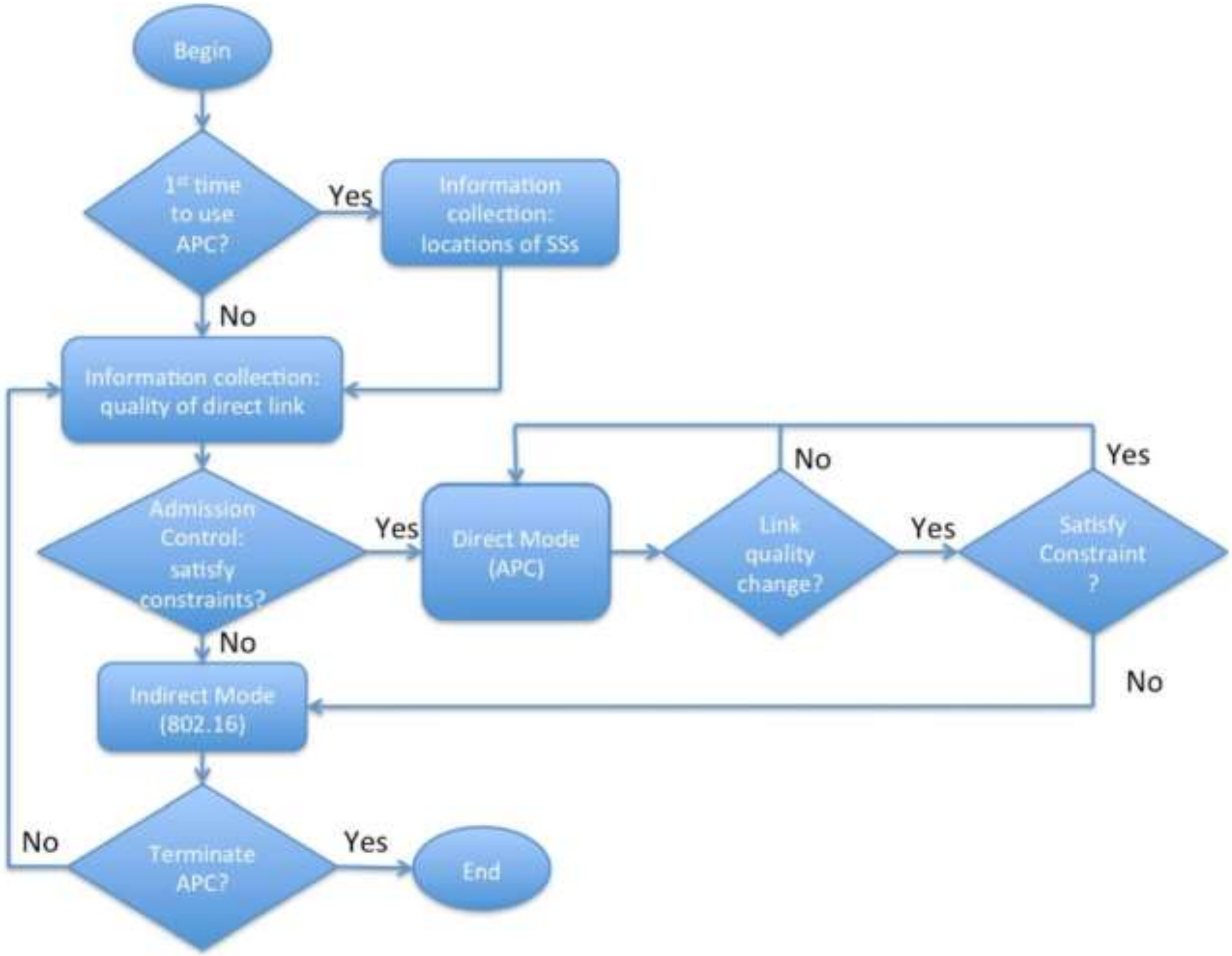

Figure .5.19: Flowchart of the switch of transmission mode.

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

Performance Comparison for 4G

The three most feasible multiple access techniques proposed for the fourth generation wireless communication systems (4G) are Orthogonal Frequency and Code Division Multiplexing (OFCDM), Orthogonal Frequency Division Multiplexing (OFDM), and Multi- Carrier Code Division Multiple Access (MC-CDMA). compares techniques in order to find the most suitable one for 4G implementation.

The transmitter structure of OFDM, MC-CDMA, OCFDM are shown in figure 5. In figure 5(a), in OFDM, the user data are viewed as a serial stream of bits and are modulated and converted into parallel sub-streams of bits. Each sub-stream is up-converted orthogonal to each other using the IDFT algorithm. At the end, cyclic prefix is added before the OFDM symbols are generated. Similarly, in figure 5(b) the MC-CDMA mode, before going into the OFDM processes, the parallel sub-streams are first spread across the frequency domain by a certain spreading factor. If before this, the user bits are also spread across the time domain as shown in the dashed square, the process becomes OCFDM.

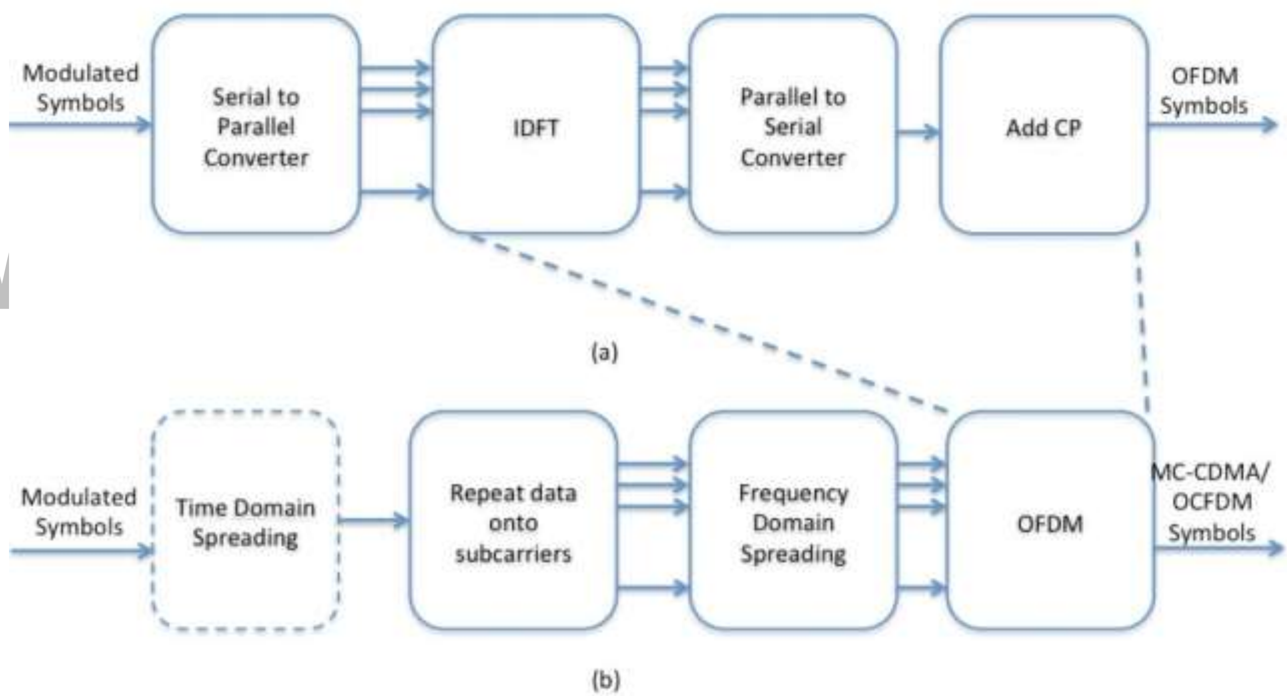

Figure 5.20: Block diagram of (a) OFDM transmitter; (b) MC-CDMA (solid line), OFCDM (dashed and solid line) transmitter.

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

To compare the performance, the modems of OFDM, MC-CDMA and OFCDM have been redesigned for fair comparison. The metric used for performance analysis is the probability of error in aspects of zero forcing and minimum mean square error. The results reveal that OFCDM provides the lowest bit error rate for a given signal noise ratio. Therefore it is considered most suitable for 4G.

Energy and Cost Efficiency

The access networks alone contribute a significant portion of the overall Internet energy consumption. With the increase of capacity demands in access networks, future-proof deployments should meet more stringent energy efficiency requirement. Traditional approaches lack energy efficiency because of their design paradigm that assumes the shortage of spectrum and the necessity of complex base stations. Therefore future approaches must be designed in an architectural level to provide inherent energy saving mechanisms.

Argument for Architectural Approaches

0.5% of the global energy consumption comes from mobile communication networks alone nowadays. Such energy consumption will be further increased in order to meet the growing demand for more capacity in BWA. The cost will increase from both deploying denser network infrastructure and the energy they will cost. Current cellular network technologies such as 3G and 4G are based on traditional design paradigms assuming spectrum shortage and the high cost of base stations. For example, 4G features high spectrum efficiency but low energy efficiency. Therefore network deployments prefer strategies with a few high-power bases stations of complex antenna systems. The blast signals through walls method used for indoor coverage is notoriously lack of energy-efficiency and is also not a sound idea considering radiation.

In the future, network deployment has to land on the tradeoff point among the cost of infrastructure, spectrum and energy. Despite the energy reduction made possible by improvements in electronics and signal processing technology, it is still not enough to match the energy consumption increasing in orders-of-magnitude because of the increasing demand in network capacity. Therefore, instead of solely increasing the efficiency of individual components, architectural solutions must be proposed in a broader scope.

Proposes a framework to analyze the total cost of network development and research some recent architectural level "clean slate" proposals that radically take advantage of newly available spectrum, energy-efficient PHY layer designs and novel backhauling strategies aiming at minimizing overall system cost. They argue that network deployment should be tightly tailored to traffic requirements by utilizing more low-power micro base stations instead of the few high power high cost base stations. Taking backhaul into consideration, a power consumption model is presented and main tradeoffs among energy, base station and spectrum cost analyzed.

Using the power consumption model, decisions can be made based on the analysis of the main characteristics of the future network infrastructure and the cost of various components it contains. The authors then draw the following conclusions from their analysis. First, not only the energy cost but also the total cost of the access network heavily depends on the number of base stations. Whereas the high cost in energy of complex base stations is inevitable, the total cost can be minimized by deploy dense micro base stations. Second, the idle power consumption and backhaul become the major factor in total cost when deploying dense base stations. Third, the energy cost also heavily depends on the availability of the spectrum. When more spectrums are made available, both the energy and the infrastructure cost can be reduced significantly by using less complex base stations.

Mixed Capacity Access

Given the fact that the optical part of WOBAN has high capacity, the capacity of the wireless access part needs to be enhanced accordingly in a low-cost fashion. Deploying multiple transceivers at each side would certainly help improve the performance. However, this will also increase the equipment and energy cost leading to low cost efficiency. Instead, if we only increase the number of transceivers at those few nodes that often experience traffic overload, the goal of increasing the capacity can be achieved without a huge increase in cost.

The Mixed-Capacity Wireless Access (MCWA) architecture for WOBAN. Under MCWA, the authors study the problem of optimally introducing a limited amount of transceivers to the wireless nodes to improve the capacity of the access network without introducing heavy cost. They model such problem as a Mixed Integer Linear Program (MILP) and later solve it using CPLEX.

An efficient channel and radio assignment can improve the utilization of MCWA by reducing interference and contention in the wireless front-end of WOBAN. also proposes one such scheme named Intelligent Channel and Radio Assignment (ICRA) for MCWA. ICRA balances the load among different channels to reduce interference as well as contention. This also can be formulated as a MILP and again solved using CPLEX.

Long Reach Access

Focuses on the intersection between energy conserving and long-reach extension technologies in BWA networks. Several architecture level approaches are reviewed on saving energy in the context of achieving long reachability. These approaches shut down idle network elements such as transceivers, shedding the rate of transmission and adaptive configuration based on current traffic. These method covers the network on the ONU side, the OLT side as well as the wireless extension side.

The researchers implement their energy saving approach in two stages: the network planning stage and the traffic engineering stage. In thefirst stage, the problem is static and an energy-efficient network planning scheme is proposed incorporating an user assignment algorithm that is behavior-aware. Under such scheme, users that have complementary traffic behaviors are assigned to the same network to achieve continuous high bandwidth utilization. Accordingly, the number of needed bandwidth and transceivers are reduced, saving a considerable amount of energy.

In the second stage, the problem is about network operations that engineer traffic dynamically. The authors propose a Dynamic Wavelength Allocation (DWA) scheme to solve the problem. As shown in figure 6, different wavelengths are assigned to Remote Nodes (RNs) and changed dynamically based the current network traffic.

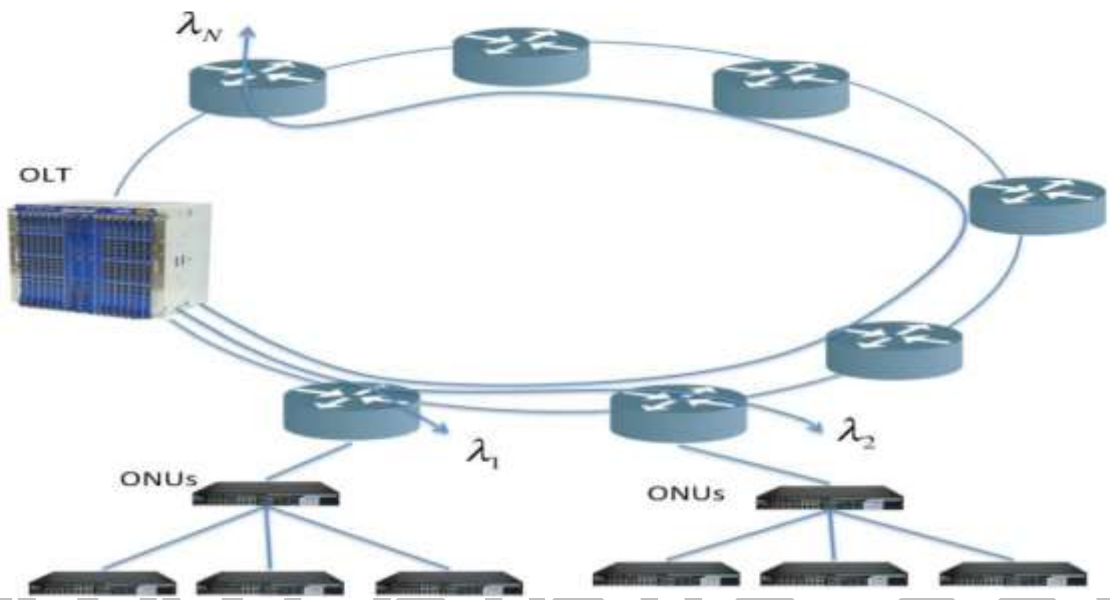

Figure 5.21: Adaptive Wavelength Allocation.

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

The authors therefore argue that extensive research attention should be paid to the study of network traffic patterns in order to support the approaches that reconfigure the network adaptively. Other energy saving approaches differentiating services according to both SLA and network utilization also provide a fair compromise between energy cost and network performance.

Green WOBAN

From previous sections we can see that the WOBAN proposal has drawn a reasonable amount of attention. Among those research efforts, dedicates its work on designing a very high-throughput and yet Green WOBAN. Novel energy saving techniques is employed to improve network utilization and energy efficiency.

As discussed before, there is an inherent capacity mismatch between the wireless front end and the optical back end of WOBAN. On one hand, the redundant capacity of the optical backhaul provides reliability guarantee in case of failure. On the other hand, such redundancy also leads to under-utilization during low-traffic hours. Regarding this discrepancy, the authors propose a coordinated ONU shut down algorithm that selectively put ONUs to sleep based on the low watermark and the high watermark threshold of the traffic profile. Thanks to the multipath feature of wireless mesh front end, user traffic can be rerouted to the active ONUs during the selective shut down.

Based on the shut down algorithm, the authors then develop a mathematical model to formulate the ONU selection problem. The model takes pre-assigned link capacities as inputs and generate a minimal set of ONUs needed to be active as output. Network parameters such as WOBAN topology and traffic are converted into constraints that can be used as guidelines for the ONU selection. With such model, a routing scheme is needed to reroute the traffic that originally take the path where the ONU is now put to sleep. In comparison to a load balancing routing scheme, the proposed routing aims at reducing network-wide energy consumption and is designed in a link state fashion. Residual capacity is assigned to a link as weight because it reflects the utilization of a link and thus indicates energy cost.

The authors use the MILP model as a benchmark to evaluate selection algorithm and the routing scheme. The impact of such energy aware design on the performance of WOBAN is analyzed over dynamic traffic profiles. The results show that appropriate configuration of design parameters can lead to considerable energy saving without significantly lowering the network performance.

IMS Architecture

The IMS (IP Multimedia Subsystem) vision is to integrate mobile/fixed voice communications and Internet technologies, bringing the power and wealth of internet services to mobile and fixed users. It allows the creation and deployment of IP-based multimedia services in the 3G networks.

IMS can enable IP interoperability for real-time services between fixed and mobile networks and so holds the promise of seamless converged voice/data services. Services transparencyand integration are key features for accelerating end-user adoption.

Two aspects of IMS are of fundamental importance to deliver these features :

• IP-based transport for both real-time and non-real-time services

• Introduction of a multimedia call model based on SIP (Session initiation Protocol).

The IMS will provide:

- A multi-service multi-protocol, multi-access, IP based network - secure, reliable andtrusted
  - Multi-services: Any type of service may be delivered by a common QoS enabled corenetwork,
  - Multi-access: diverse access networks (WiFi, WiMAX, UMTS, CDMA2000, xDSL,Cable, etc.) can interface with IMS

IMS is Not one network, but different networks that interoperate seamlessly thanks toroaming agreements between any type of IMS service provider.

IMS is « An enabler » for Service Providers to offer :

- Real-time and non real-time, communication services between peers, or in a client-serverconfiguration.
- Mobility of services and mobility of users (Nomadicity)
- Multiple sessions and services simultaneously over the same connection. One can gethis communications services anywhere, on any terminal.

Layers of the IMS Architecture

The IMS architecture as defined by the 3GPP standards is an all-packet core network that creates an access-agnostic environment to deliver a wide range of multimedia services that auser can access using any device or network connection. Leveraging the SIP protocol, IMS supports IP-to-IP sessions over any wireline connection (e.g., DSL, cable) or wireless network protocol (e.g., Wi-Fi, GSM or CDMA). The IMS infrastructure allows a carrier to interwork between the TDM and IP networks to provide a seamless service experience.

Access layer : IMS is access independent. In case of mobile, it can be GPRS, EDGE (also called enhanced GPRS), UMTS or Wireless LAN. 3GPP UMTS R5 focuses on EDGE and UMTS accesses. 3GPP UMTS R6 adds WLAN. 3GPP2 assumes cdma2000 accesses. Fixed service providers will apply IMS to ADSL and cable network accesses.
Transport layer : It is an all-IP network that consists of IP routers (edge and core IP routers).

Connectivity layer = Access Layer and Transport Layer

Session Control layer : Comprises network control servers for managing calls or establishing sessions and modifications. The two main elements of this layer are the CSCF (call session control function) and the HSS (home subscriber server). Sometimes called the SIP server, the CSCF performs end-point registration and routing of the SIP signalingmessages to the application server related to a particular service. In addition, the CSCF interworks with the access and transport layers to guarantee QoS for all services. The HSS database maintains each end user's service profile. Stored in a central location, this information could include location information, service triggers, etc.
Application layer : Utilizes application and content servers to provide various value-added services. At the heart of this layer are the AS (application server), MRFC (multimedia resource function controller), and the MRFP (multimedia resource function processor). The AS is responsible for the execution of service-specific logic, for example call flows and user interface interactions with subscribers, while the

MRFP—more commonly known as the IP media server—provides adjunct media processing for the application layer. Through the media server, a service provider can deliver various non-telephony services (e.g., push-to- talk) as well as speech-enabled services, video services, and other more mainstream services such as conferencing, prepaid card and personalized ring-back tones.

Control and application layers are access and transport independent so that a user can access to his/her IMS services from any access.

Underlying Concepts of the IMS Architecture

A set of requirements has been introduced for the design of IMS :

- **IP connectivity**

A fundamental requirement is that a client has to have IP connectivity to access IMSservices. In addition, it is required that IPv6 is used.

- **Access Independence**

The IMS is designed to be access-independent so that IMS Services can be provided over any IP connectivity networks (e.g., GPRS, WLAN, broadband access xDSL, etc). Release 5 IMS specifications contain some GPRS-specific features. In Release 6 (e.g., GPRS) access- specific issues are separated from the core IMS description.

- **Ensures quality of service from IP Multimedia Services**

Via the IMS, the terminal negotiates its capabilities and expresses its QoS requirements during a Session Initiation Protocol (SIP) session set-up or session modification procedure. The terminal is able to negotiate such parameters as: Media type, Media type bit rate, packetsize, packet transport frequency, bandwidth, etc. After negotiating the parameters at the application level, the terminals reserve suitable resources from the access network. When end-to-end QoS is created, the terminals encode and packetize individual media types with an appropriate protocol (e.g., RTP) and send these media packets to the access andtransport network by using a transport layer protocol (e.g., TCP or UDP) over IP.

- **IP Policy control for ensuring correct usage of media resources**

IP policy control means the capability to authorize and control the usage of bearer traffic intended for IMS media, based on the signaling parameters at the IMS session. This requiresinteraction between the IP connectivity access network and the IMS.

- **Secure communication**

The IMS provides at least a similar level of security as the corresponding GPRS and GSM networks. The IMS ensures that users are authenticated before they can start using services,and users are able to request privacy when engaged in a session.

- **Charging arrangements**

The IMS architecture allows different charging capabilities to be used, particularly, off-line (postpaid) and on-line (prepaid) charging.

- **Support of roaming**

The roaming feature makes it possible to use services even though the user is not geographically located in the service area of the home network.

- **Interworking with other networks**

To be a new, successful communication network technology and architecture, the IMS has tobe able to connect to as many users as possible. Therefore, the IMS supports communication with PSTN, ISDN, mobile and Internet users. Additionally, it will be possibleto support sessions with Internet applications that have been developed outside the 3GPP community.

- **Service control**

IMS provides all the network with all the information about the services the user has subscribed to, so that standardized mechanisms are used to enable the network invoking theuser's services.

- **Service development**

IMS provides service capabilities for multimedia service development. Presence, Conferencing, instant messaging, push-to-talk are examples of service capabilities.

IMS and SIP

SIP is an application-layer protocol defined by IETF (RFC 3261) that can establish, modify, and terminate multimedia sessions (conferences) over the Internet. A multimedia session is aset of senders and receivers and the data streams owing from the senders to the receivers. For example, a session may be a telephony call between two parties or a conference call among more than two parties. SIP can also be used to invite a participant to an ongoing session such as a conference. SIP messages could contain session descriptions such that participants can negotiate with media types and other parameters of the session. SIP provides its own mechanisms for reliable transmission and can run over several different transport protocols such as TCP, UDP, and SCTP (Stream Control Transmission Protocol). SIP is also compatible with both IPv4 and IPv6. SIP provides the following key capabilities formanaging multimedia communications:

- Determine destination user 's current location.
- Determine whether a user is willing to participate in a session.
- Determine the capabilities of a user 's terminal.
- Set up a session.
- Manage a session. This includes modifying the parameters of a session, invoking servicefunctions to provide services to a session, and terminating a session.

Like HTTP, SIP is a client-server protocol that uses a request and response transaction model. A SIP client is any network element that generates SIP requests and receives SIP responses. A SIP server is a network element that receives SIP requests in order to service them and sends back responses to these requests.

IMS Network Entities

IMS Terminal

It is an application on the user equipment that sends and receives SIP requests. It representsa software on a PC, on an IP phone or on a UMTS mobile station (UE, User Equipment).

Home Subscriber Server (HSS)

The Home Subscriber Server (HSS) is the main data storage for all subscriber and service- related data of the IMS. The main data stored in the HSS include user identities, registrationinformation, access parameters and service-triggering information

Call State Control Function (CSCF)

In GSM, a user can roam on to visited networks provided that the visited network can accessthe home HLR and an agreement exists between the two operators. The same kind of roaming is supported for R5 multimedia services. In GSM roaming, call control always takes place in the visited network, the only connection to the home network being access to the HLR.

In IMS, there was a long and complicated discussion about whether IP multimedia call control for roamers should take place in the visited or home network. Those who said it should take place in the home network pushed the argument that the user would have signedfor a range of services, and many of these would not be available or would work differently in a visited network. Those who favored visited network control were concerned about the long delays and signaling traffic created by having all services controlled from the home network that might be located on a different continent.

In the end, it was decided that IMS control would be controlled from the home network. This complication gives rise to three flavors of CSCF (Proxy CSCF, Interrogating, Serving CSCF).CSCF = Call Stateful Control Function.

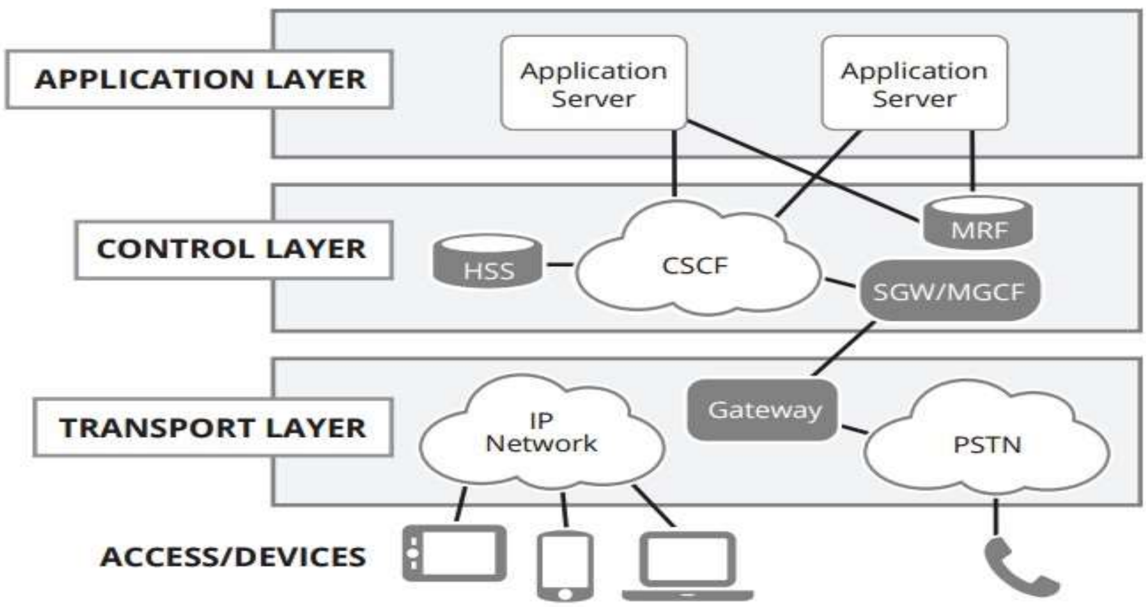

Fig.5.11: IMS Architecture

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

A **P-CSCF** (Proxy CSCF) is a mobile's first contact point inside a local (or visited) IMS.

It acts as a SIP Proxy Server. In other words, the P-CSCF accepts SIP requests from themobiles and then either serves these requests internally or forwards them to other servers.

The P-CSCF includes a Policy Control Function (PCF) that controls the policy regarding howbearers in the GGSN should be used.

The P-CSCF performs the following specific functions :

• Forward SIP REGISTER request from a mobile to the mobile 's home network.

• Forward other SIP messages from a mobile to a SIP server (e.g., the mobile's S-CSCF inthe mobile's home network).

• Forward SIP messages from the network to a mobile.

• Perform necessary modifications to the SIP requests before forwarding them to othernetwork entities.

• Maintain a security association with the mobile.

• Detect emergency session.

• Create CDRs.

An **I-CSCF** (Interrogating CSCF) is a contact point within an operator's network for all connections destined to a subscriber of that network operator. There may be multiple I- CSCFs within an operator's network. The functions performed by the I-CSCF are :

• To contact the HSS to obtain the name of the S-CSCF that is serving a user

• To assign an S-CSCF based on received capabilities from the HSS. An S-CSCF is assigned if there is no S-CSCF allocated.

• To forward SIP requests or responses to the S-CSCF

• To provide a hiding functionality.

• The I-CSCF is an optional function that can be used to hide an operator network 's internal structure from an external network when an I-CSCF is used.

An **S-CSCF** (Serving CSCF) provides session control services for a user. It maintains session states for a registered user 's on-going sessions and performs the following main tasks :

• Registration : An S-CSCF can act as a SIP Registrar to accept users' SIP registration requests and make users 'registration and location information available to location servers such as the HSS (Home Subscriber Server).

• Session Control : An S-CSCF can perform SIP session control functions for a registered user. It relays SIP requests and responses between calling and called parties.

• Proxy Server : An S-CSCF may act as a SIP Proxy Server that relays SIP messages between users and other CSCFs or SIP servers.

• Interactions with Application Servers : An S-CSCF acts as the interface to application servers and other IP or legacy service platforms.

• Other functions : An S-CSCF performs a range of other functions not mentioned above. For example, it provides service-related event notifications to users and generates Call Detail Records (CDRs) needed for accounting and billing.

Before being able to use the services of the IMS domain such as establish a multimedia session or receive a session request, a user should register to the network. Wherever the user is (home or visited network), this registration procedure involves a P-CSCF. All the signaling messages sent and received by the user terminal are forwarded by the P-CSCF. The terminal never knows the I-CSCF or S-CSCF addresses.

MGCF, IMS-MGW and T-SGW : Interworking with PSTN/GSM

The IMS networks need to interact with PSTN so that IMS users can establish services to PSTN users. The architecture for supporting PSTN and legacy mobile networks is shown in Figure 2. The interworking between IMS networks and PSTN/legacy networks occur at two levels: One is the user plane level and the other is the signaling plane level. In the user plane, interworking elements are required to convert IP based media streams on the IMS side to TDM based media streams on the PSTN side. The **IMS-Media Gateway** (IMS-MGW) element is responsible for this function. The IS-MGW elements are controlled by the **Media Gateway Control Function** (MGCF) through the Megaco protocol. On the signaling plane level, the SIP signaling needs to be converted to legacy signaling such as ISDN Signaling User Part (ISUP). The MGCF is responsible for converting SIP signaling to legacy signaling such as ISUP. The MGCF is responsible for transporting ISUP signaling messages to a Trunking Signaling Gateway (T-SGW) over IP transport bearer. The T-SGW transports these ISUP messages over the SS7 bearer to either the PSTN or the legacy wireless networks. Please note that MGCF and T-SGW are logical functions. These functions may be implemented in one physical box.

The PSTN switch reserves a voice circuit among those it shares with the IMS-MGW and sends an ISUP IAM message over SS7 to a T-SGW (Trunking Signaling Gateway). The T- SGW is responsible for signaling transport conversion. It forwards the ISUP IAM message to the MGCF entity over SIGTRAN (Signaling Transport over IP).

The MGCF creates a context in the IMS-MGW using the MEGACO/H.248 protocol. This context consists of an association between a TDM termination and an RTP termination. The TDM termination terminates the voice circuit the IMS-MGW shares with the PSTN switch. The RTP termination terminates the RTP channels between the IMS-MGW and the IMS terminal.

The MGCF entity originates a SIP INVITE method containing the SDP description returnedby the IMS-MGW. This method is sends to the IMS subsystem that delivers it to thedestination IMS terminal.

• A virtual network operator that deploys IMS and relies on the access networks of third partyoperators.

These operators can deploy their own IMS services and open their service architecture to ASP that interface with their own application servers through the ISC interface.

Moreover, There are different approaches for the deployment of an IMS based-infrastructure for a service provider with mobile and fixed infrastructures.

• It may deploy **two independent IMS-based infrastructures** for fixed (e.g., xDSL) and mobile (e.g., EDGE or UMTS). The equipment of these two infrastructures may be from the same equipment vendor.

- It may deploy **a unique IMS-based infrastructure** from a vendor that is capable of supporting the two types of accesses.
- It may consider **a hybrid IMS-based infrastructure** with two IMS network infrastructures but sharing common application servers.

IMS Service Architecture

The IMS service architecture consists of Application Servers (AS), Multimedia Resource Functions (MRF) also called IP media servers, and S-CSCF equivalent to call servers

The **AS** (Application Server) provides a service execution environment, application-specific logic (e.g., Push To Talk, Presence, Prepaid, Instant messaging), and all the signaling forone or more services. It may influence and impact the SIP session on behalf of the services. The AS corresponds to the SCF (Service Control Function) of the Intelligent Network.

The **MRF** (Multimedia Resource Function) is a network element whose sole purpose is the processing of media streams, also known as RTP streams for network-based services. Media streamprocessing includes such functions as playing announcements, collecting DTMF digits, audio recording and playback, bridging multiple streams (also known as conferencing), speech recognition, text-to-speech rendering, and video processing. In performing these functions, the MRF's role in the network is that of a slave device: it always operates under the direct control of an AS. The MRF corresponds to the SRF (Specialized Resource function) of the Intelligent Network.

The call server called **S-CSCF** (Serving - Call State Control Function) plays the role of a point from which services are invoked. It has the user profile that indicates the services the user has subscribed to, and the conditions under which services are invoked. The S-CSCF corresponds to the SSF (Service Switching function) of the Intelligent Network.

Entities of the IMS Service Architecture

The IMS services architecture allows deployment of new services by operators and third party service providers. This provides subscribes a wide choice of services. The

S-CSCF is the anchor point for delivering new services since it manages the SIP sessions. However, services can be developed and deployed in a distributed architecture. Multiple service platforms may be used to deploy wide variety of services. The IMS defines three different ways of delivering services :

• **Native SIP Services**: In the last few years, a wide variety of technologies have been developed by various organizations for developing SIP services. They include SIP servlets, Call Processing Language (CPL) script, SIP Common Gateway Interface (CGI) and Java APIs for Integrated Networks (JAIN). One or more **SIP Application Servers** may be used to deploy services using these technologies. It is intended to allow the SIP Application Server toinfluence and impact the SIP session on behalf of the services. Service Control Interaction Manager (SCIM) is a specialized type of SIP Application Server, that performs the role of interaction management between other application servers.

• **Legacy IN services:** While new and innovative services are required, the legacy telephony services cannot be ignored provided by CAMEL. The **IMS Service Switching Point** (IMS- SSP) is a particular type of application server the purpose of which is to host the CAMEL network features (i.e. trigger detection points, CAMEL Service Switching Finite State Machine, etc) and to interface to CSE (CAMEL Service Environment). The IM SSP and the CAP interface support legacy services only.

• **Third party services:** UMTS has defined Open Services Access (OSA)7 to allow 3 rd partyservice providers to offer services through UMTS network. The OSA offers a secure API for third party service providers to access UMTS networks. Therefore, subscribers are not restricted to the services offered by the operators. **OSA Service Capability Server** (OSA SCS) interfaces to the OSA framework Application Server and provides a standardized way

for third party secure access to the IM subsystem. The OSA reference architecture defines an OSA Application Server as an entity that provides the service logic execution environmentfor client applications using the OSA API. This definition of Application Server differs from thedefinition of Application Server in the context of service provisioning for the IM subsystem, i.e. the entity communicating to the S-CSCF via the ISC interface.

In addition the Application Servers can also interact with the **Multimedia Resource Function controller** (MRFC) via the S-CSCF in order to control Multimedia Resource Function processing (MRFP). MRF = MRFC+MRFP.

All the Application Servers, (including the IM-SSF and the OSA SCS) behave as SIP application server on their interface towards the S-CSCF.

Deployment of an IMS Architecture

From the above concepts, one can state that :

• IMS is access independent. GPRS/EDGE/UMTS users as well as fixed broadband users(xDSL, cable) can access to IMS.

• IMS provides the interface towards the circuit switched networks (e.g., PSTN, GSM).

• IMS provides a standardized interface (ISC, IMS Service Control) to access to services.

IMS can be deployed by :

• A mobile service provider to offer advanced services and multimedia services to its usersGPRS/EDGE/UMTS.

• A fixed network operator (xDSL, cable)

INTRODUCTION ABOUT 4G

4G stands for Forth Generation of Cellular Communications and is the next step in the evolution of mobile data. 4G provides high mobility with high speed data rates and also supports high capacity IP-based services and applications while it also maintains full backward compatibility.

It is also based on wireless communication that is IP based and is slated on Advanced MIMO technology. 4G technologies follow Multiple Input Multiple Output Technology that uses signal multiplexing between multiple transmitting antennas (space multiplex) and time or frequency.

Fourth generation (4G) technology will offer many advancements to the wireless market, including downlink data rates well over 100 megabits per second (Mbps), low latency, very efficient spectrum use and low-cost implementations.

4G enhancements promise to bring the wireless experience to an entirely new level with impressive user applications, such as sophisticated graphical user interfaces, high-end gaming, high-definition video and high-performance imaging.

4G VISIONS

The 4G systems are designed to provide a wide variety of new services, from high-quality voice to high definition video to high-data-rate wireless channels. The term 4G is used broadly to include several types of BWA communication systems, not only cellular systems. 4G is described as MAGIC — Mobile multimedia, anytime anywhere, Global mobility support, integrated wireless solution, and customized personal service.

The 4G systems will not only support the next generation mobile services, But also will support the fixed wireless networks. The 4G systems are about seamlessly integrating terminals, networks, and applications to satisfy increasing user demands.

Accessing information anywhere, anytime, with a seamless connection to a wide range of information and services, and receiving a large volume of information, data, pictures, video, and so on, are the keys of the 4G infrastructure.

The future 4G systems will consist of a set of various networks using IP as a common protocol. 4G systems will have broader bandwidth, higher data rate, and smoother and quicker handoff and will focus on ensuring seamless service across a multiple of wireless systems and networks.

The key is to integrate the 4Gcapabilities with all the existing mobile technologies through the advanced techniques of digital communications and networking. Application adaptability and being highly dynamic are the main features of 4G services of interest to users. These features mean services can be delivered and be available to the personal preference of different users and support the users' traffic, air interfaces, radio environment, and quality of service. Connection with the network applications can be transferred into various forms and levels correctly and efficiently.

The following figure illustrates elements and techniques to support the adaptability of the 4G domain. The fourth generation will encompass all systems from various networks, public to private; operator-driven broadband networks to personal areas; and ad hoc networks. The 4G systems will interoperate with 2G and 3G systems, as well as with digital (broadband) broad casting systems.

In addition, 4G systems will be fully IP-based wireless Internet. This all-encompassing integrated perspective shows the broad range of systems that the fourth generation intends to integrate, from satellite broadband to high altitude platform to cellular 3G and 3G systems to WLL (wireless local loop) and FWA (fixed wireless access) to WLAN (wireless local area network) and PAN (personal area network),all with IP as the integrating mechanism. With 4G, a range of new services and models will be available. These services and models need to be further examined for their interface with the design of 4G systems.

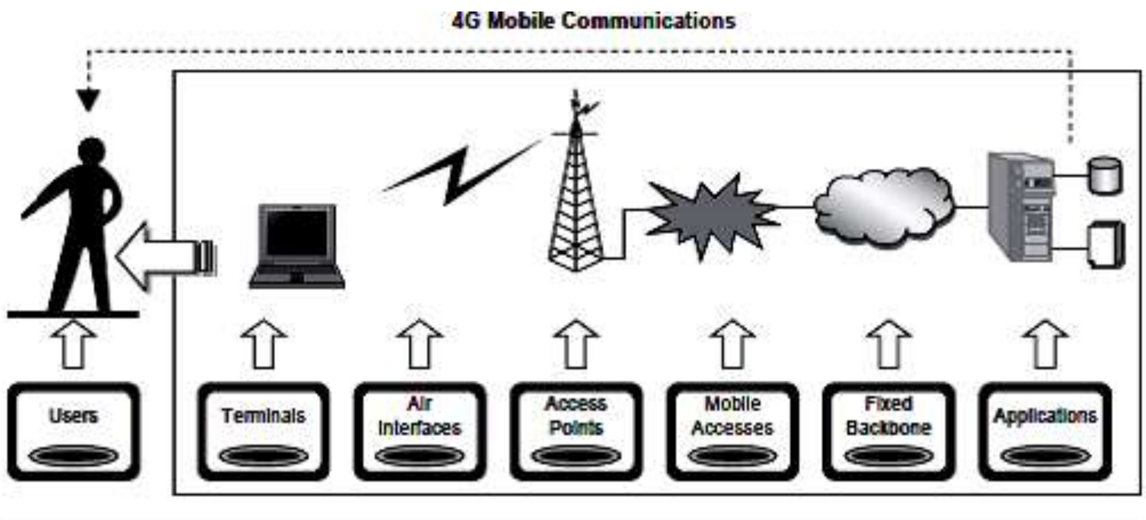

Fig.5.1: 4G Visions

[Source: Text book- Wireless Communications and network First Edition, Elsevier 2007 by Vijay Garg]

Table 5.1: Comparison of key parameters of 4G with 3G.

Details	3G including 2.5G (EDGE)	4G
Major requirement driving Architecture	Predominantly voice driven, data was always add on	Converge data and voice over IP
Network architecture	Wide area cell-based	Hybrid-integration of WLAN (WiFi, Bluetooth) and wireless wide-area networks
Speeds	384 kbps to 2 Mbps	20 to 100 Mbps in mobile mode
Frequency band	Dependent on country or continent (1.8 to 2.4 GHz)	Higher frequency bands(2 to 8 GHz)
Bandwidth	5 to 20 MHz	100 MHz or more
Switching design basis	Circuit and packet	All digital with packetized
Access technologies	WCDMA, CDMA2000	OFDM and multicarrier
Forward error correction	Convolutional codes rate 1/2, 1/3	Concatenated coding
Component design	Optimized antennadesign, multiband adapters	Smart antenna, software defined multiband and wideband radios
Internet protocol(IP)	Number of air link protocol including IPv5.0	All IP (IPv6.0)
Mobile top speed	200 km/h	200 km/h

4G will need to be highly dynamic in terms of support for:

The users' traffic

Air interfaces and terminal types

Radio environments

Quality-of-service types

Mobility patterns.

4G FEATURES

Some key features of 4G mobile networks are as follows.

High usability: anytime, anywhere, and with any technology

Support for multimedia services at low transmission cost

Personalization

Integrated services

Support for interactive multimedia, voice, streaming video, Internet, and other broadband services

IP based mobile system

High speed, high capacity, and low cost per bit

Global access, service portability, and scalable mobile services

Seamless switching, and a variety of Quality of Service driven services

Better scheduling and call admission control techniques

Ad hoc and multi hop networks

Better spectral efficiency

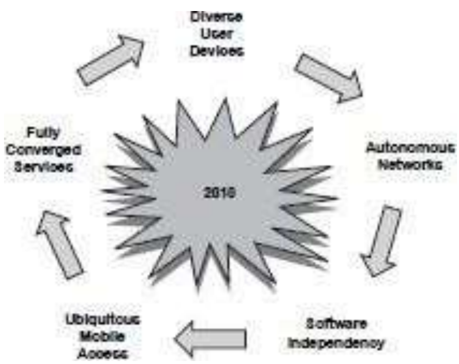

Fig5.2: 4G Features

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier

2007 by Vijay Garg]

4G networks will be all-IP-based heterogeneous networks that will allow users to use any system at anytime and anywhere. Users carrying an integrated terminal can use a wide range of applications provided by multiple wireless networks. 4G systems will provide not only telecommunications services, but also data and multimedia services. To support multimedia services, high-data- rate services with system reliability will be provided. At the same time, a low per-bit transmission cost will be maintained by an improved spectral efficiency of the system.

Personalized service will be provided by 4G networks. It is expected that when 4G services are launched, users in widely different locations, occupations, and economic classes will use the services. In order to meet the demands of these diverse users, service providers will design personal and customized service for them. 4G systems will also provide facilities for integrated services. Users can use multiple services from any service provider at the same time.

4G technologies are significant because users joining the network add mobile routers to the network infrastructure. Because users carry much of the network with them, network capacity and coverage is dynamically shifted to accommodate changing user patterns. Users will automatically hop away from congested routes to less congested routes. This permits the network to dynamically and automatically self-balance capacity, and increase network utilization.

In a cellular infrastructure, user’s network contribution is nil. They are just consumers competing for resources. But in wireless ad hoc peer-to-peer networks, users cooperate - rather than compete - for network resources.

LTE

LTE stands for Long Term Evolution and it was started as a project in 2004 by telecommunication body known as the Third Generation Partnership Project (3GPP). SAE (System Architecture Evolution) is the corresponding evolution of the GPRS/3G packet core network evolution. The term LTE is typically used to represent both LTE and SAE.

LTE evolved from an earlier 3GPP system known as the Universal Mobile Telecommunication System (UMTS), which in turn evolved from the Global System for Mobile Communications (GSM). Even related specifications were formally known as the evolved UMTS terrestrial radio access (E-UTRA) and evolved UMTS terrestrial radio access network (E-UTRAN).

A rapid increase of mobile data usage and emergence of new applications such as MMOG (Multimedia Online Gaming), mobile TV, Web 2.0, streaming contents have motivated the 3rd Generation Partnership Project (3GPP) to work on the Long-Term Evolution (LTE) on the way towards fourth-generation mobile.

The main goal of LTE is to provide a high data rate, low latency and packet optimized radio access technology supporting flexible bandwidth deployments. Same time its network architecture has been designed with the goal to support packet-switched traffic with seamless mobility and great quality of service.

Facts about LTE

- LTE is the successor technology not only of UMTS but also of CDMA2000.
- LTE bring up to 50 times performance improvement and much better spectral efficiency to cellular networks.
- LTE introduced to get higher data rates, 300Mbps peak downlink and 75 Mbps

Peak uplink. In a 20MHz carrier, data rates beyond 300Mbps can be achieved under very good signal conditions.

- LTE is an ideal technology to support high date rates for the services such as voice over IP (VOIP), streaming multimedia, videoconferencing or even a high-speed cellular modem.

- LTE uses both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) mode. In FDD uplink and downlink transmission used different frequency, while in TDD both uplink and downlink use the same carrier and are separated in Time.

- LTE supports flexible carrier bandwidths, from 1.4 MHz up to 20 MHz as well as both FDD and TDD. LTE designed with a scalable carrier bandwidth from 1.4 MHz up to 20 MHz which bandwidth is used depends on the frequency bandnetwork operator.

- All LTE devices have to support (MIMO) Multiple Input Multiple Output transmissions, which allow the base station to transmit several data streams over the same carrier simultaneously.
- All interfaces between network nodes in LTE are now IP based, including the backhaul connection to the radio base stations. This is great simplification compared to earlier technologies that were initially based on E1/T1, ATM and frame relay links, with most of them being narrowband and expensive.

- Quality of Service (QoS) mechanism have been standardized on all interfaces to ensure that the requirement of voice calls for a constant delay and bandwidth, can still be met when capacity limits are reached.

- Works with GSM/EDGE/UMTS systems utilizing existing 2G and 3G spectrum and new spectrum. Supports hand-over and roaming to existing mobile networks.

Advantages of LTE

- **High throughput:** High data rates can be achieved in both downlink as well as uplink. This causes high throughput.
- **Low latency:** Time required to connect to the network is in range of a few hundred milliseconds and power saving states can now be entered and exited very quickly.
- **FDD and TDD in the same platform**: Frequency Division Duplex (FDD) and Time Division Duplex (FDD), both schemes can be used on same platform.
- **Superior end-user experience:** Optimized signaling for connection establishment and other air interface and mobility management procedures have further improved the user experience. Reduced latency (to 10 mms) for better user experience.
- **Seamless Connection**: LTE will also support seamless connection to existing networks such as GSM, CDMA and WCDMA.
- **Plug and play:** The user does not have to manually install drivers for the device. Instead system automatically recognizes the device, loads new drivers for the hardware if needed, and begins to work with the newly connected device.
- **Simple architecture:** Because of Simple architecture low operating expenditure (OPEX).

LTE Network Architecture

The high-level network architecture of LTE is comprised of following three main components:

- The User Equipment (UE).

- The Evolved UMTS Terrestrial Radio Access Network (E-UTRAN).
- The Evolved Packet Core (EPC).

The evolved packet core communicates with packet data networks in the outside world such as the internet, private corporate networks or the IP multimedia subsystem. The interfaces between the different parts of the system are denoted Up, S1 and GI as shown below:

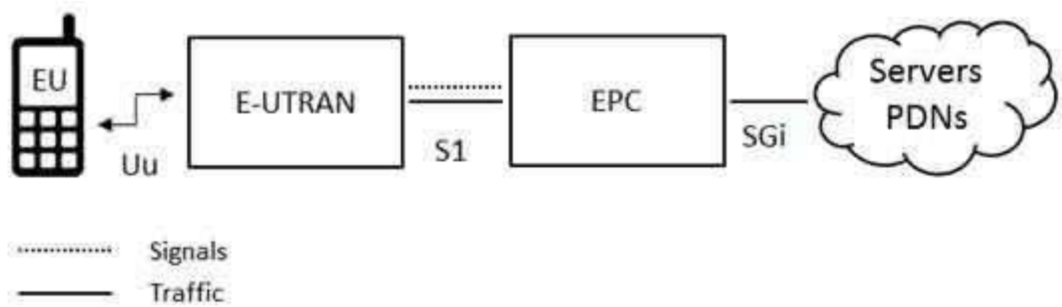

Fig.5.12: The LTE network Architecture

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Garg]

The User Equipment (UE):

The internal architecture of the user equipment for LTE is identical to the one used by UMTS and GSM which is actually a Mobile Equipment (ME). The mobile equipment comprised of the following important modules:

- **Mobile Termination (MT)** :This handles all the communication functions.
- **Terminal Equipment (TE)** : This terminates the data streams.
- **Universal Integrated Circuit Card (UICC)** :This is also known as the SIM card for LTE equipment's. It runs an application known as the Universal Subscriber Identity Module (USIM).

A **USIM** stores user-specific data very similar to 3G SIM card. This keeps information about the user's phone number, home network identity and security keys etc.

The E-UTRAN (The access network)

The architecture of evolved UMTS Terrestrial Radio Access Network (E-UTRAN) has been illustrated below.

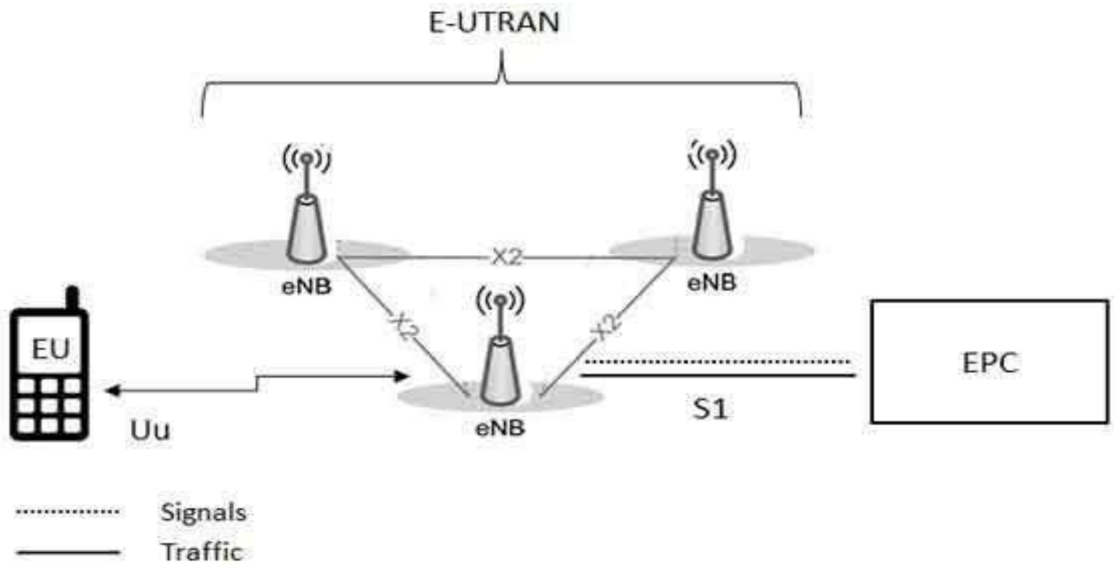

Fig.5.13 The architecture of E-UTRAN

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Garg]

The E-UTRAN handles the radio communications between the mobile and the evolved packet core and just has one component, the evolved base stations, called anode or eNB. Each eNB is a base station that controls the mobiles in one or more cells. The base station that is communicating with a mobile is known as its serving end.

LTE Mobile communicates with just one base station and one cell at a time and there are following two main functions supported by eNB:

- The eBN sends and receives radio transmissions to all the mobiles using the analogue and digital signal processing functions of the LTE air interface.
- The eNB controls the low-level operation of all its mobiles, by sending them signaling messages such as handover commands.

Each eBN connects with the EPC by means of the S1 interface and it can also be connected to nearby base stations by the X2 interface, which is mainly used for signalling and packet forwarding during handover.

A home eNB (HeNB) is a base station that has been purchased by a user to provide femtocell coverage within the home. A home eNB belongs to a closed subscriber group (CSG) and can only be accessed by mobiles with a USIM that also belongs to the closed subscriber group.

The Evolved Packet Core (EPC) (The core network):

The architecture of Evolved Packet Core (EPC) has been illustrated below. There are few more components which have not been shown in the diagram to keep it simple. These components are like the Earthquake and Tsunami Warning System (ETWS), the Equipment Identity Register (EIR) and Policy Control and Charging Rules Function (PCRF).

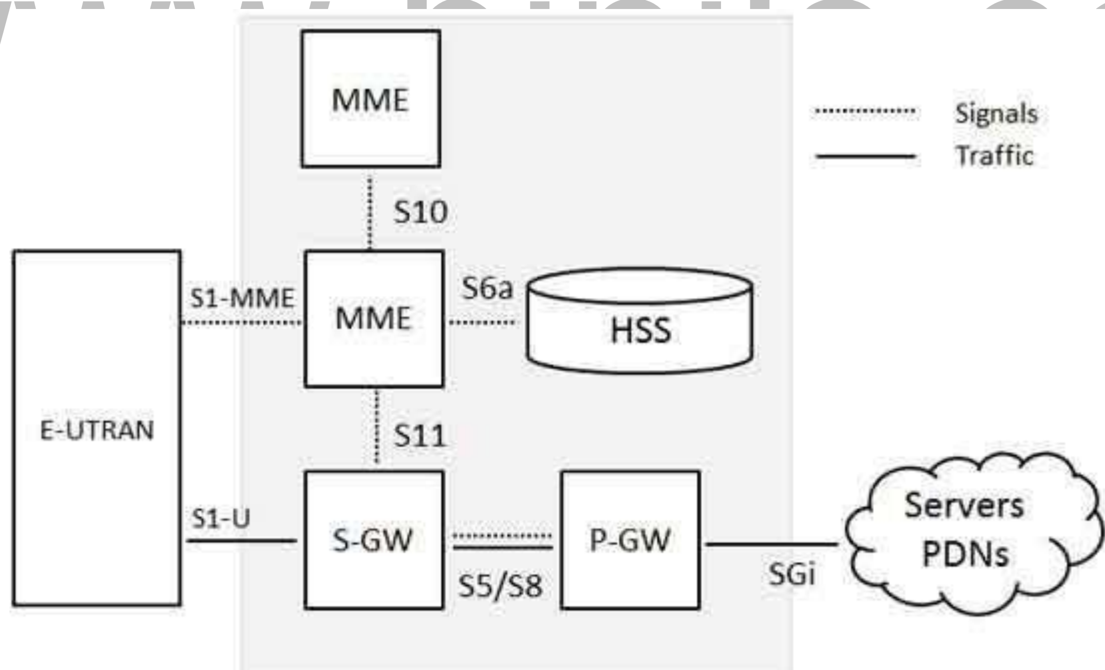

Fig.5.14: The architecture of Evolved Packet Core

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Garg]

Below is a brief description of each of the components shown in the above architecture?

- The Home Subscriber Server (HSS) component has been carried forward from UMTS and GSM and is a central database that contains information about all the network operator's subscribers.

- The Packet Data Network (PDN) Gateway (P-GW) communicates with the outside world ie. Packet data networks PDN, using SGi interface. Each packet data network is identified by an access point name (APN). The PDN gateway has the same role as the GPRS support node (GGSN) and the serving GPRS support node (SGSN) with UMTS and GSM.

- The serving gateway (S-GW) acts as a router, and forwards data between the base station and the PDN gateway.

- The mobility management entity (MME) controls the high-level operation of the mobile by means of signaling messages and Home Subscriber Server.

- The Policy Control and Charging Rules Function (PCRF) is a component which is not shown in the above diagram but it is responsible for policy control decision-making, as well as for controlling the flow-based charging functionalities in the Policy Control Enforcement Function (PCEF), which resides in the P-GW.

The interface between the serving and PDN gateways is known as S5/S8. This has two slightly different implementations, namely S5 if the two devices are in the same network, and S8 if they are in different networks.

2G/3G versus LTE

Following table compares various important Network Elements & Signaling protocols used in 2G/3G and LTE.

2G/3G	LTE
GERAN and UTRAN	E-UTRAN
SGSN/PDSN-FA	S-GW
GGSN/PDSN-HA	PDN-GW
HLR/AAA	HSS
VLR	MME
SS7-MAP/ANSI-41/RADIUS	Diameter
DiameterGTPc-v0 and v1	GTPc-v2
MIP	PMIP

LTE Protocol Stack Layers

The below diagram shows the layers available in E-UTRAN Protocol Stack.

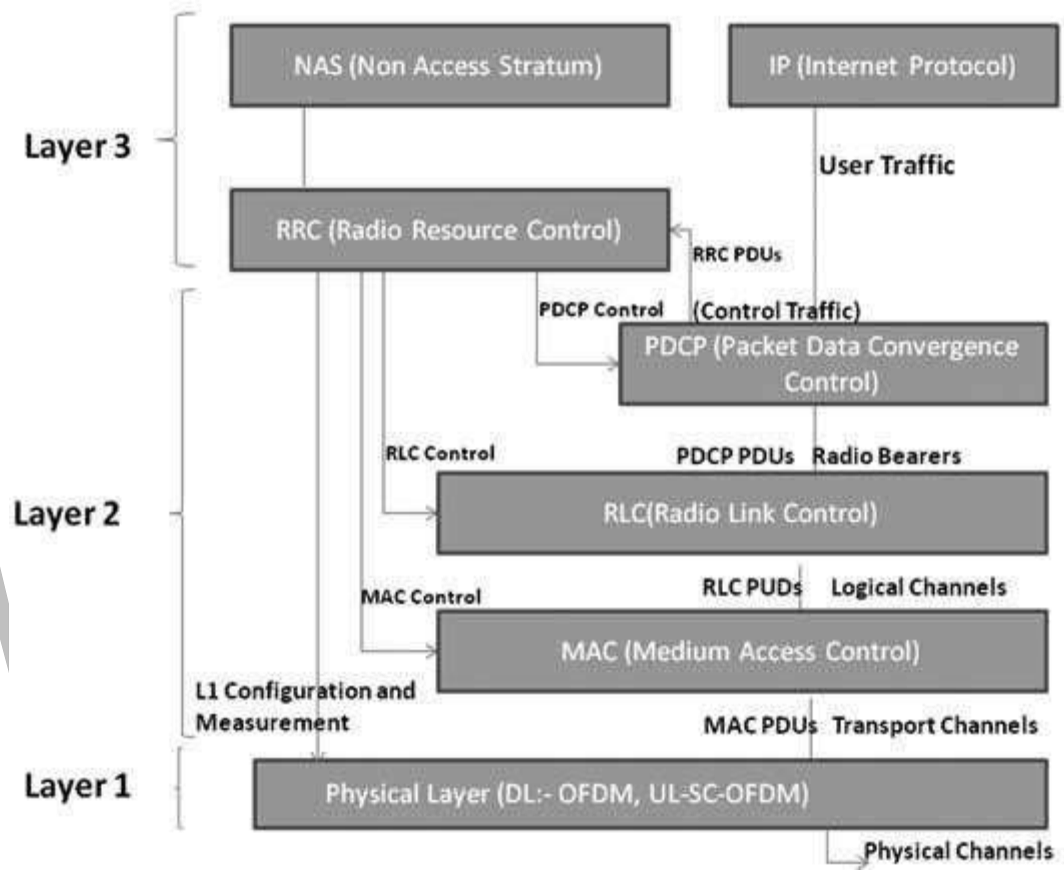

Fig. 5.15 E-UTRAN Protocol Stack

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Garg]

Physical Layer(Layer 1)

Physical Layer carries all information from the MAC transport channels over the air interface. Takes care of the link adaptation (AMC), power control, cell search (for initial synchronization and handover purposes) and other measurements (inside the LTE system and between systems) for the RRC layer.

Medium Access Layer(MAC)

MAC layer is responsible for Mapping between logical channels and transport channels, Multiplexing of MAC SDUs from one or different logical channels onto transport blocks (TB) to be delivered to the physical layer on transport channels, DE multiplexing of MAC SDUs from one or different logical channels from transport blocks (TB) delivered from the physical layer on transport channels, Scheduling information reporting, Error correction through HARQ Priority handling between UEs by means of dynamic scheduling, Priority handling between logical channels of one UE, Logical Channel prioritization.

Radio Link Control (RLC)

RLC operates in 3 modes of operation: Transparent Mode (TM), Unacknowledged Mode (UM), and Acknowledged Mode (AM).

RLC Layer is responsible for transfer of upper layer PDUs, error correction through ARQ (Only for AM data transfer), Concatenation, segmentation and reassembly of RLC SDUs (Only for UM and AM data transfer).

RLC is also responsible for re-segmentation of RLC data PDUs (Only for AM data transfer), reordering of RLC data PDUs (Only for UM and AM data transfer),duplicate detection (Only for UM and AM data transfer), RLC SDU discard (Only for UM and AM data transfer), RLC re-establishment, and protocol error detection(Only for AM data transfer).

Radio Resource Control (RRC)

The main services and functions of the RRC sub layer include broadcast of System Information related to the non-access stratum (NAS), broadcast of System Information related to the access stratum (AS), Paging, establishment, maintenance and release of an RRC connection between the UE and E-UTRAN, Security functions including key

Management, establishment, configuration, maintenance and release of point to point Radio Bearers.

Packet Data Convergence Control (PDCP)

PDCP Layer is responsible for Header compression and decompression of IP data, Transfer of data (user plane or control plane), Maintenance of PDCP Sequence Numbers (SNs), In-sequence delivery of upper layer PDUs at re-establishment of lower layers, Duplicate elimination of lower layer SDUs at re-establishment of lower layers for radio bearers mapped on RLC AM, Ciphering and deciphering of user plane data and control plane data, Integrity protection and integrity verification of control plane data, Timer based discard, duplicate discarding, PDCP is used for SRBs and DRBs mapped on DCCH and DTCH type of logical channels.

Non Access Stratum(NAS) Protocols

The non-access stratum (NAS) protocols form the highest stratum of the control plane between the user equipment (UE) and MME.NAS protocols support the mobility of the UE and the session management procedures to establish and maintain IP connectivity between the UE and a PDN GW.

Mobile Virtual Network Operator (MVNO)

An MVNO is a business model that emerges when the traditional mobile value chain is ruptured. Therefore, new players can participate in the mobile value chain and extract value leveraging their valuable assets.

The traditional mobile value chain can be separated into two main areas: 1.-Radio access network that is exclusively exploited by mobile network operators, moreover it requires a license granted by the regulatory authority to use the spectrum, and 2.-the rest of the elements required to de liver the service to the customers. As it is shown in the exhibit 1, this second area of the value chain includes: the operation of the core network(e.g. switching, backbone, transportation, etc.), the operation of the value added services (e.g. SMS, voicemail, etc.), the operation of the back office process to support business process (e.g. subscriber registration, handset and SIM logistic, billing, balance check, top-up network, customer care, etc.), the definition of a mobile value offer and the final delivery of the products and services to the client through the distribution channel. It is in this second area of the value chain where other parties can participate by innovating, operating or selling mobile services.

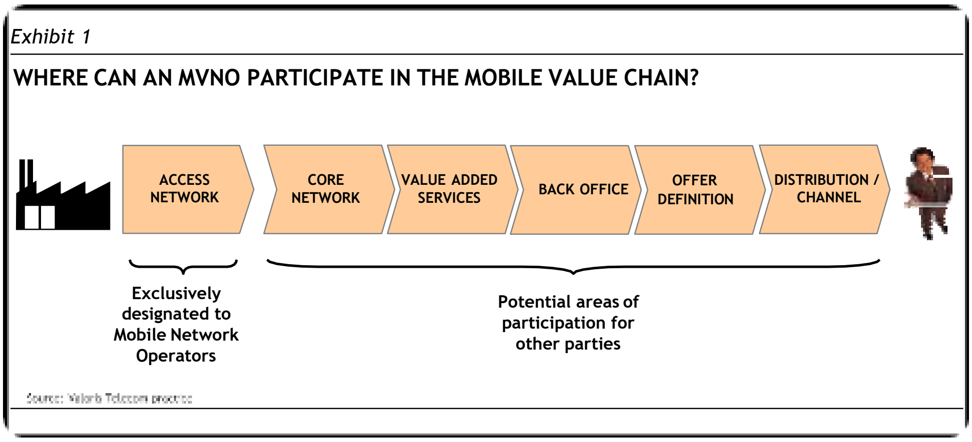

Figure 5.22: Mobile Virtual Network Operator

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

Both Mobile Network Operators (MNOs) and Mobile Virtual Network Operators (MVNOs) can take advantage of this business model. Mobile network operators can exploit its network capacity, IT infrastructure and service and product portfolio to acquire untargeted segments, add a new revenue stream from wholesale business and reduce spare capacity and cost-to-serve per user. On the other hand, an MVNO can exploit its brand awareness, distribution channels and customer base to provide customized value proposition and complementary products and services to its customer An MVNO venture brings multiple benefits to a company such as: a new revenue stream, a low-cost entry strategy to the mobile market, a new vehicle to strengthen the value proposition and an opportunity to increase customer acquisition and/or retention.

MVNO business models

The different business models in the MVNO market are based on how the value chain is restructured. Therefore, four main business models that emerge are: Branded Reseller, Light-MVNO, Full-MVNO and Network enablers.

Branded reseller is the lightest MVNO business model, where the venture just provides its brand and, sometime, its distribution channels. While the mobile network operator (MNO) provides the rest of the business, from access network to the definition of the mobile service offer. This is the model that requires the lowest investment for a new venture, therefore the fastest to implement. However, most of the business levers remain with then network provider (MNO or MVNE). Therefore, the new venture has a very limited control of the business levers and value proposition of the service.

Full-MVNO is the most complete model for a new venture, where the mobile network operator just provides the access network infrastructure and, sometimes, part of the core network, while the new venture provides the rest of the elements of the value chain. This MVNO business model is typically adopted by telecom players that could gain synergies from their current business operation.

Light-MVNO is an intermediate model between a branded reseller and a full-MVNO. This model allows new ventures to take control of the marketing and sales areas and, in some cases, increase the level of control over the back-office processes and valued-added services definition and operations.

Network enablers, typically known as Mobile Virtual Network Enablers (MVNE), this is a third party provider focused on the provision of infrastructure that facilitate the launch of MVNO operations. An MVNE can be positioned between a host MNO and an MVNO venture to provide service ranging from value added services and back office processes to offer definition. MVNEs reduce the entry barriers of MVNO ventures, given that an MVNE aggregates the demand of small players to negotiate better terms and conditions with host MNO. They pass on some of these benefits to their MVNO partners. Moreover, the all- in-a-box approach to launch an MVNO through a MVNE has accelerated, even more, the explosion of the MVNO market. Some MVNE models are also called Mobile Virtual Network

Aggregator (MVNA), depending on the range of services offered or whether they aggregate different host MNOs. MVNE models range from telco-in-a-box offering, where the MVNE just offers core network, value added services and back office services, to full MVNE as shown in the exhibit 2.

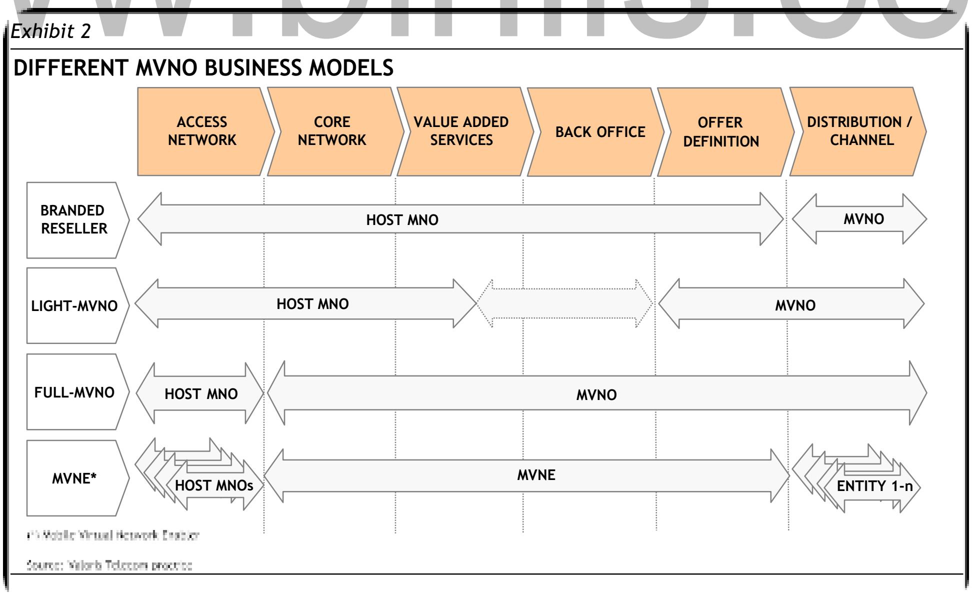

Figure 5.23: **MVNO business models**

[Source: Text book- Wireless Communications and networking , First Edition, Elsevier 2007 by Vijay Garg]

Players in the MVNO arena

There are three groups of players that have taken advantages of the MVNO business model: Telecom companies, non-telecom companies and investors.

Mobile network operators can benefit from the MVNO model by serving untapped segments that their current value proposition is unable to attract. Additionally, telecom operators in general (fixed and/or mobile) can use the MVNO opportunity to enter to new geographies through a wholesale business based on a MVNE model. Finally, telecom operators with no mobile offering can strengthen its value proposition providing a 4-play offer.

Non-telecom companies such as retailers, media and content generators, financial institutions, travel and leisure, postal services, sport clubs, food and beverage, among other type of companies can take advantage of their current assets (brand, customer base, channels, content, etc.) to exploit a new business or to strengthen their current value proposition and customer loyalty.

Investors can take advantage of the MVNO model by investing in new opportunities to create value by participating in the telecom industry.

Key enablers for take-off and benefits of the MVNOs

The MVNO business model requires a series of conditions to be in place for take-off. For instance, the level of readiness of the MVNO market players (e.g. mobile network operators) is also crucial for this opportunity.

Therefore, if the mobile network operators see the MVNOs as a potential threat instead of a new way to extract more value form the market, their willingness to participate will be lower. The main enablers of the MVNO market include:

- Level of commitment of the regulator agency to increase competition through the implementation of the MVNO business model.
- Mobile network operator cooperation, having a strategy and see it as win-win situation. Therefore, network, IT and processes readiness of the mobile network operators to provide wholesale services to third parties.

Moreover, there are specific conditions that could determine when it is the right time to impulse or pursue the MVNO business model in a certain mobile market

Most of them are related to the available room for growth in the sector in terms of subscribers and level of competition. Among the various indicators that mobile markets have shown before the launch of MVNOs, the main ones are:

- High level of mobile penetration (higher than 80-90%), therefore the available room to expand the market is reducing and operators will grow through the acquisition of customers from its competitors.
- Market growth has slowed down, most of the time it has happened because the level of competition has gone down.
- There are 3-4 mobile network operators that have already invested in and deployed an extensive infrastructure. Therefore, there is enough capacity that has led to spare capacity in some of the competitors.
- Low competition environment in the mobile market coupled with a low level of customer satisfaction that has led to high level of churn.
- High level of prices and low level of innovation due to significant market power of current players that has led to a low level of competition.

There are number of benefits behind the MVNO business model that have been obtained in the markets where the business model has been launched:

- Market growth stimulation by serving untapped segments.
- Open competition to avoid oligopolies, reducing entry barriers to new players. This situation has led to intensify competition resulting in: greater choice of service providers and services, price decreases that has benefited customers and a wave of innovative value propositions and advanced services.
- Improvement in service quality.
- Stimulate private and foreign investment that acts as a new source of employment and economic growth.

MVNO business evolution

MVNO has emerged in almost every continent. Europe and U.S. are the markets that have seen develop more MVNOs ventures than rest of the world However, it is in Europe where most of world's MVNOs are operational and having success. Moreover, they have reached more than 20% of market share in certain European markets such as UK and Belgium. Additionally, the geographical spread of the MVNO business model has reached southern Europe, where the model has been rapidly expanding towards countries such as Spain, Italy, Portugal and, more recently, Turkey. However there are other areas of the world where the model has been gaining a relevant presence. For instance, Asia, especially in the Middle East, has adopted this model with countries such as Jordan and UAE that have already established the conditions to launch MVNOs. Finally, Latin America is also starting to adopt the model in countries like Chile, where the regulatory agency has created a legal framework and has granted around 15 licenses to operate MVNOs.

The MVNO business cycle has an evolution similar to any other industry: starting with a nascent stage where few players are present in the market, followed by explosion stage where many players emerge and steady growth is seen, and finally, a consolidation phase where the number of players is reduced. For instance, if we take the MVNO landscape in Europe, we can see that there are different levels of market maturity. For instance, Portugal is clearly in the nascent phase, this market is taking its first steps after many years of stagnation promoted by MNOs. Spain, on the other hand, is experiencing the explosion phase. The Spanish market has rapidly grown, from no MVNO activity in 1Q 2007 to around 30 players one and half years later. Finally, Sweden is in the consolidation phase, where some players have acquired their competitors during the last years.

Where are the Opport unitie The main business rational to launch an MVNO can be summarized in: 1.- Access to current customer base, 2.- Ability to deliver a differentiated value proposition and 3.- Potential synergies with existing business. Therefore, the

introduction of the MVNO model brings opportunities to different types of players that can put together one or more of these elements. Companies that are able to use them will earn the right to play the MVNO game.

Moreover, a good execution will guarantee a successful performance for thenew venture.

Companies that have a captive customer base are able to develop an MVNO business. These companies can lever not only its brands awareness among their customers, but also their distribution channels to sell mobile servicesto their current customers. These companies should consider the mobile service as a new product of their current product portfolio or, inclusively, insome cases, as a new private label. Additionally, companies that serve or sell products to niche market are able to develop an MVNO business. These companies normally have an exclusive and non-replicable content and value added services that can strengthen the mobile service value proposition.

Finally, companies that can use an MVNO as an extension of existing value proposition (e.g. a Telco from 3-play to 4-play). Additionally, companies can lever their existing IT and core network infrastructure to exploit the MVNO opportunity.

The MVNO business model can bring different types of opportunity depending on the assets your company can exploit. Moreover, the type of opportunity that your company is able to exploit depends on the industry it belongs to.

- If you are a retailer you can offer mobile services through an MVNO as a new product category or as part of a loyalty program (e.g. Tesco Mobile in UK and Ireland and Carrefour Mobile throughout Europe).
- If you are an MNO you can expand in your home market with a second brand through an MVNO to serve new segments (e.g. UZO in Portugal by MTN) or expand to foreign markets through an MVNO and/or MVNE offering (e.g. KPN in Spain and France as a MVNE and with its own MVNO brands such as Sim yo and blau).

- If you are a Fixed/Broadband operator you can offer mobile services through an MVNO as a step to build 4-play and convergent solutions (e.g. TELE2 throughout Europe and Ono in Spain).
- If you are a long distance operator you can offer mobile services through an MVNO to immigrant segments that would benefit from low-rate international calls.
- If you are in media or other industries, such as sport clubs, with a strong brand and community you can offer mobile services through an MVNO as a new business line to increase the value of your clients leveraging your current positioning (e.g. M6 in France and Football Club Porto in Portugal).
- If you are a financial service institution you can offer mobile services through an MVNO as a new distribution and communication channel or as part of a retention program (e.g. Bank inter in Spain and Rabo bank in Netherlands).
- If you are an investor, the MVNO market represents a new investment opportunity in a nascent market or a value generation source through M&A transactions in markets in the process of consolidation (e.g. Planet Ventures in the Middle East and Penta Investments in Central Europe).

SMART ANTENNA TECHNIQUES

Smart antenna techniques, such as multiple-input multiple-output (MIMO) systems, can extend the capabilities of the 3G and 4G systems to provide customers with increased data throughput for mobile high-speed data applications. MIMO systems use multiple antennas at both the transmitter and the receiver to increase the capacity of the wireless. With MIMO systems, it may be possible to provide in excess of 1 Mbps for 2.5G wireless TDMA EDGE and as high as 20 Mbps for 4G systems. With four antennas at the transmitter and receiver, this has the potential to provide four times the data rate of a single antenna system without an increase in transmit power or bandwidth.

MIMO techniques can support multiple independent channels in the same bandwidth, provided the multipath environment is rich enough. The number of trans mitting antennas is M , and the number of receiving antennas is N , where N ≥M.

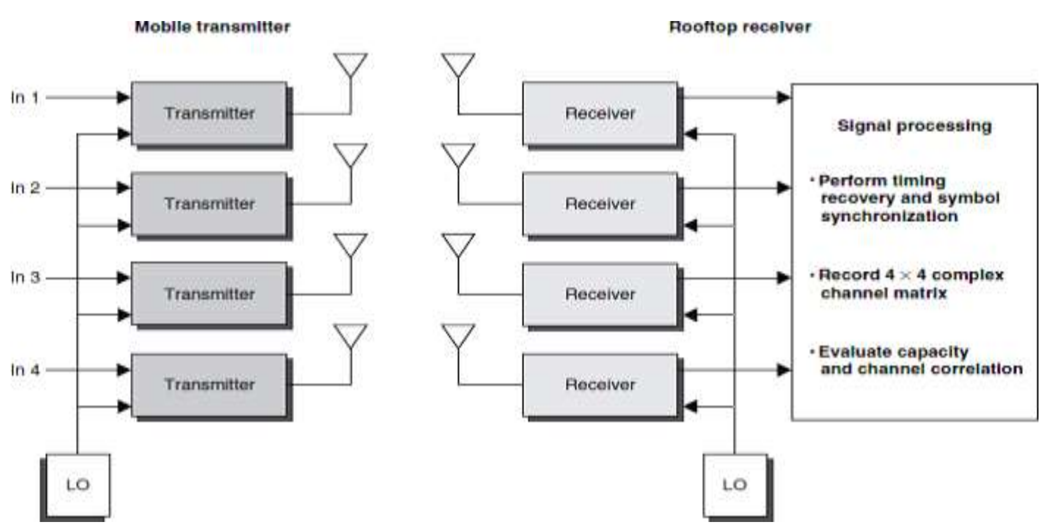

Fig.5.5: Smart Antenna

There are four cases:

- Single-input, single-output (SISO);
- Single-input, multiple-output (SIMO);
- Multiple-input, single-output (MISO);
- Multiple-input, multiple-output (MIMO).

MIMO – SISO (Single Input Single Output)

The simplest form of radio link can be defined in MIMO terms as SISO – Single Input Single Output. This is effectively a standard radio channel – this transmitter operates with one antenna as does the receiver. There is no diversity and no additional processing required.

Fig.5.6: SISO - Single Input Single Output

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Gag]

The advantage of a SISO system is its simplicity. SISO requires no processing in terms of the various forms of diversity that may be used. However the SISO channel is limited in its performance as interference and fading will impact the system more than a MIMO system using some form of diversity. The throughput depends upon the channel bandwidth and the

Signal to noise ratio.

If the channel bandwidth is B, the transmitter power is P_t, the signal at the receiver has an average SNR of SNR_0, and then the Shannon limit on channel capacity C is

$$C \approx B \log_2 (1 + SNR_0)$$

MIMO – SIMO (Single Input Multiple Output)

The SIMO or Single Input Multiple Output version of MIMO occurs where the transmitter has a single antenna and the receiver has multiple antennas. This is also known as receiving diversity. It is often used to enable a receiver system that receives signals from a number of independent sources to struggle the effects of fading.

SIMO has the advantage that it is relatively easy to implement. The use of SIMO may be quite acceptable in many applications, but where the receiver is located in a mobile device such as a cellphone handset, the levels of processing may be limited by size, cost and battery drain.

There are two forms of SIMO that can be used:

Maximum ratio combining SIMO:

This form of SIMO takes both signals and sums them to give the combination. In this way, the signals from both antennas contribute to the overall signal.

Fig.5.7: SIMO - Single Input Multiple Output

There are N antennas at the receiver. If the signals received on the antennas have on average the same amplitude, then they can be added coherently to produce an N 2 increase in signal power. There are N sets of noise sources that are added coherently and result in an N –fold increase in noise power. The capacity for this channel is approximately equal to

$$C \approx B \log2 (1 + N \times SNR0)$$

MIMO – MISO (Multiple Input Single Output)

Multiple Input Single Output (MISO) is also termed transmit diversity. In this case, the same data is transmitted redundantly from the two transmitter antennas. The receiver is then able to receive the optimum signal which it can then use to receive extract the required data.

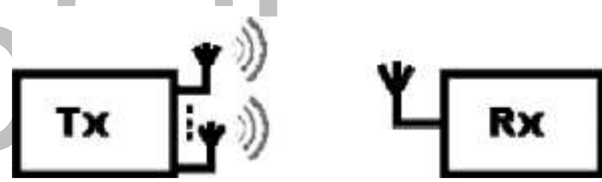

Fig.5.8: MISO - Multiple Input Single Output

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Gag]

It has M transmitting antennas. The total power is divided into M transmitter branches. If the signals add coherently at the receiving antenna, we get an M - fold increase in SNR as compared to SISO. Because there is only one receiving antenna, the noise level is the same as SISO. The overall increase in SNR is approximately.

MIMO (Multiple Input Multiple Output)

MIMO is effectively a radio antenna technology as it uses multiple antennas at the transmitter and receiver to enable a variety of signal paths to carry the

Data, choosing separate paths for each antenna to enable multiple signal paths to be used.

The two main formats for MIMO are given below:

- **Spatial diversity:**

Spatial diversity used in this narrower sense often refers to transmit and receive diversity. These two methodologies are used to provide improvements in the signal to noise ratio and they are characterized by improving the reliability of the system with respect to the various forms of fading.

- **Spatial multiplexing:**

This form of MIMO is used to provide additional data capacity by utilizing the different paths to carry additional traffic, i.e. increasing the data throughput capability.

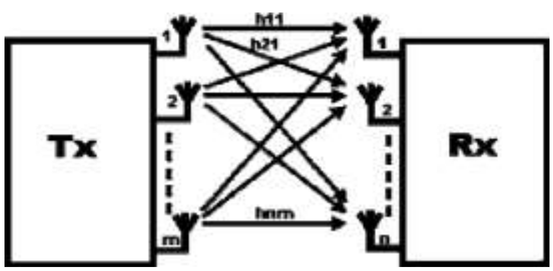

Fig.5.9: MIMO - Multiple Input Multiple Output

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Gag]

MIMO systems can be viewed as a combination of MISO and SIMO channels. In this case, it is possible to achieve approximately an MN -fold increase in the average SNR 0 giving a channel capacity equal to

$C \approx B \log2 (1 + M \times N \times SNR0)$

OFDM MIMO SYSTEMS

Multiple-input multiple-output (MIMO) wireless technology in combination with orthogonal frequency division multiplexing (MIMOOFDM) is an air- interface solution for next-generation wireless local area networks (WLANs), wireless metropolitan area networks (WMANs), and fourth-generation mobile cellular wireless systems. OFDM and MIMO techniques can be combined to achieve high spectral efficiency and increased throughput.

For high-data rate transmissions, the MIMO channel is frequency selective (multipath). OFDM can transform such channel into a set of parallel frequency-flat channels (reduce Rx complexity).

MIMO-OFDM Transmitter

The Source bit stream is encoded by the FEC encoder and the coded bit stream mapped to a constellation by digital modulator, and encoded by the MIMO encoder. Each of the parallel output symbol streams are corresponding to a certain Transmitting antenna follows the same Transmitting process:

- Insertion of pilot symbols (synchronization)
- Modulation by inverse FFT
- Attachment of CP and Preamble

Finally, the data frame is transferred to IF/RF stage for Transmitter

MIMO-OFDM Receiver

The received symbol stream from different Receiving antennas is fist synchronized Preambles and CPs must be extracted from Received symbol stream. The Remaining OFDM symbols demodulated by FFT. Frequency pilots are extracted from the demodulated OFDM symbols, and are used for channel estimation. Estimated channel matrix aids the

MIMO decoder and the estimated Transmitted symbols are demodulated and then decoded. All MIMO- OFDM receiver must perform time synchronization; frequency offset estimation, and correction and parameter estimation. This is generally carried out using a preamble consisting of one or more training sequences. Once the acquisition phase is over, receiver goes into the tracking mode.

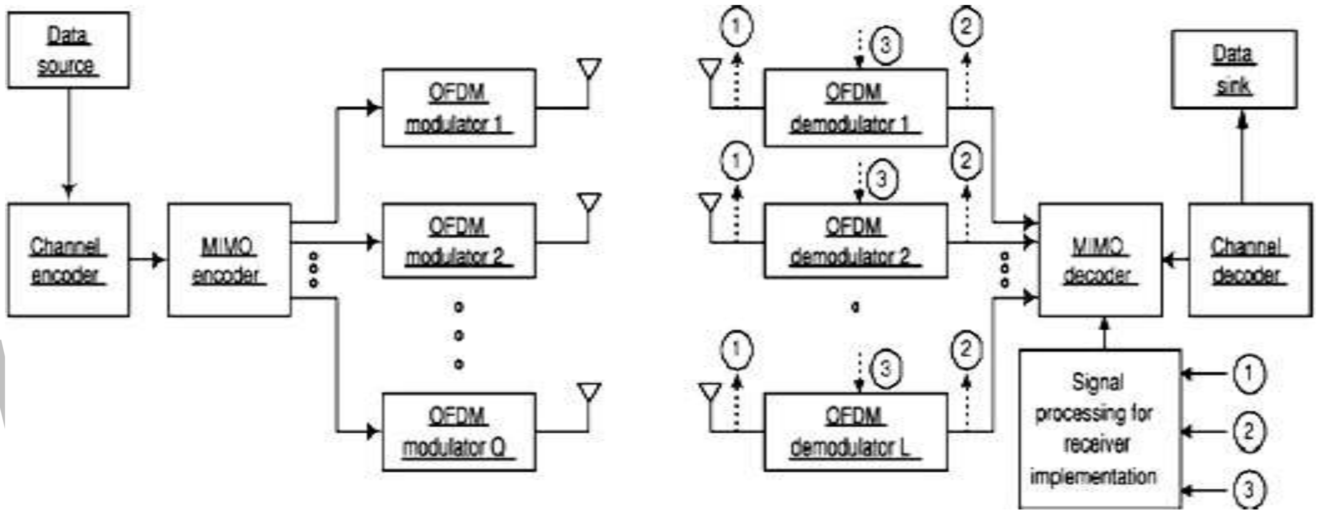

Fig.5.9: Block Diagram MIMO – OFDM Frame Structure

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Gag]

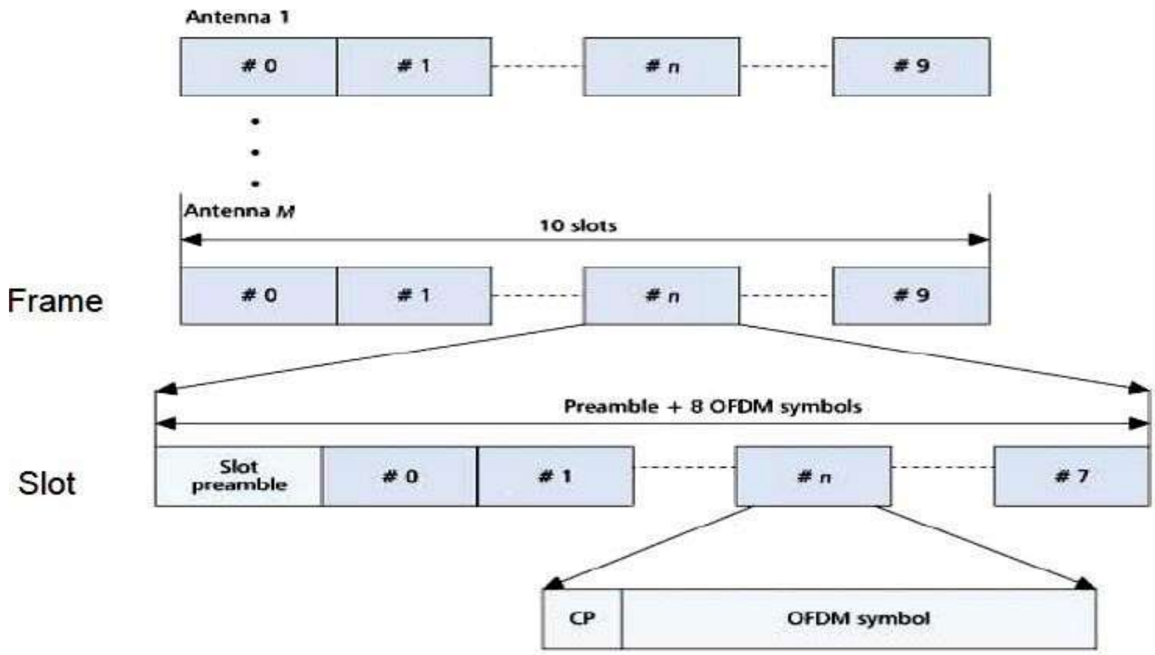

Fig.5.10 MIMO – OFDM Frame Structure

[Source: Text book- Wireless Communications and networking, First Edition, Elsevier 2007 by Vijay Garg]

In the time domain, a frame is a minimum transmission unit that includes 10 slots. Each slot consists of 1 slot preamble and 8 OFDM symbols. The preamble is used for time synchronization. Each OFDM symbol in a slot is attached to a CP that is used to reduce ISI and simplify channel equalizer. The frame is structured such that data and pilot symbols are transmitted over subcarriers (timing phase, timing frequency, and frequency offset estimation)

SIGNALING IN MIMO-OFDM SYSTEMS

The signaling schemes used in MIMO systems can be grouped into spatial multiplexing which realizes capacity gain, and space-time coding, which improves link reliability through diversity gain. Most multi-antenna signaling schemes, in fact, realize both spatial-multiplexing and diversity gain. A framework for characterizing the trade-off between spatial-multiplexing and diversity gains in flat-fading MIMO channels.Spatial Multiplexing multiplexes multiple spatial channels to send as many independent data as possible over different antennas.

There are four types of spatial multiplexing schemes:

- Diagonal BLAST
- Horizontal BLAST
- V-BLAST and
- Turbo BLAST

The method to estimate Transmitting signals has three steps:

- Estimate the channel matrix (training sequence)
- Determine optimal detecting order and nulling vectors
- Detect the received signals based on optimal detecting order and successive interference cancellation

SPATIAL MULTIPLEXING IN MIMO-OFDM SYSTEMS

In an OFDM-based MIMO system, spatial multiplexing is performed by transmitting data streams on a tone-by-tone basis with the total transmit power split uniformly across antennas and tones. Although the use of OFDM eliminates ISI, the computational complexity of MIMO-OFDM spatial- multiplexing receivers can still be high. Computational complexity reductions are

Attained by performing channel inversion in the case of a minimum mean- squared error (MMSE) receiver on a subset of tones only and computing the remaining inverses or QR factors, respectively, through interpolation.

MIMO – OFDM STANDARDS

1. IEEE 802.11n (MIMO) Systems
2. IEEE 802.11a (OFDM) Systems
3. IEEE 802.11a & g (WLAN) Systems

4. IEEE 802.11a & g (WMAN) Systems
5. IEEE 802.16a (Wi MAX) Systems

Advantage of OFDM – MIMO systems are:

1. High spectral efficiency& capacity
2. Simple implementation by FFT (fast Fourier transform);
3. Low receiver complexity;
4. Robust ability for high-data-rate transmission over multipath fading channel
5. High flexibility in terms of link adaptation
6. Low complexity multiple access schemes such as orthogonal frequency division multiple access.

Applications of OFDM – MIMO systems are:

- Wireless network
- Next generation network(4G)
- Wi-Fi, Wi-MAX, W – MAN
- Digital TV
- Power line control
- Digital audio and video broadcasting
- Discrete Multitude systems