East West University

"Performance Enhancement & Optimization of GSM System through the Analysis of Key Performance Indicator for QoS"

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A thesis submitted in partial fulfillment of the requirements for the degree of Bachelor of Science in Electronics and Telecommunication Engineering.

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Declaration

This report on the basis of our thesis paper and its enhancement of studies throughout our thesis work is submitted to follow the terms and conditions of the department of Electronics and Communications Engineering. This report is the requirement for the successive competition of B.Sc. Engineering in Electronics and Communications Engineering.

We state that the report along with its literature that has been demonstrated in this report paper is our own work with the masterly guidance and fruitful assistance of our supervisor for the finalization of our report successfully.

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Acknowledgement

We would like to extend our gratitude to the many people who helped to bring this project to fruition. First, we would like to thank Assistant Professor Mustafa Mahmud Hussain for his help, professionalism, valuable guidance and throughout this project and through my entire program of study that we do not have enough words to express our deep and sincere appreciation.

Finally, we must express our very profound gratitude to our parents and to our faculty members of ECE Department for providing us with unfailing support and continuous encouragement throughout our years of study and through the process of researching and writing this thesis. This accomplishment would not have been possible without them.

Thank you all for your grate support to us.

Approval

This thesis report “Performance Enhancement & Optimization of GSM System through the Analysis of Key Performance Indicator for QoS” submitted by Shanjeda Amir, ID: 2012-1-55-010 and Musharath Khan, ID: 2012-1-56-008 to department of Electronics and Communications Engineering ,East West University has been accepted as satisfactory for the partial fulfillment of the requirements for the degree of Bachelor of Science in Electronics and Communication Engineering and approved as to its style and content.

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Abstract

In this modern era of technology the field of telecommunication has advanced tremendously worldwide, enhancing it an ever expanding process which not only revolutionized the arena of telecommunication but also accelerated the process of making the world a step closer of being a single "Global Village". In addition different technical advancement has made the existing mobile companies to upgrade their various services continuously in order to sustain in this competitive world.

Quality of service (QoS) is an important **key performance indicator** (KPI) that is used in determining the efficiency of an industry in terms of services rendered. In telecommunication system, accessibility, reliability and connection (voice) quality are three major factors used in evaluating quality of service of an operator.

And for consumers in the industry, it is expected that maximum satisfaction be derived from any services paid for.

“Performance Enhancement & Optimization of GSM System through the Analysis of Key Performance Indicator for QoS”

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Performance Enhancement & Optimization of GSM System through the Analysis of Key Performance Indicator for Quality Of Service

Chapter 1

Introduction :

Quality of Service or QoS is an important factor that enhances the mechanisms which include the performance control, stability, reliability, scalability, usability and customer satisfaction. Therefore not only it has to be standardized but also needs to be monitored whether necessary changes in the specifications are required.

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The term (maximum satisfaction) has now become a difficult task to achieve especially in GSM industry. One of the major reasons attributable to this is as a result of mismatch between expansion in customer base and infrastructural (Network) expansion. In this report we standardize the benchmark of different performance parameters involved in a calling process for the mobile operators to ensure a quality service in Bangladesh.
To obtain the quality of service in order to improve the network infrastructure we have to understand the basic elements of mobile communications.

Bangladesh is a rapidly growth country in terms of Mobile Subscribers. Till March 2016, the number of mobile subscribers were **130.881 m**illion [1] and the number is rapidly increasing. To increase the subscribers in Bangladesh, Mobile Network Operators are improving their network infrastructure .

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OPERATOR	SUBSCRIBER (IN MILLIONS)
Grameen Phone Ltd. (GP)	56.285
Banglalink Digital Communications Limited	31.932
Robi Axiata Limited (Robi)	27.450
Airtel Bangladesh Limited (Airtel)	10.161
Pacific Bangladesh Telecom Limited (Citycell)	0.799
Teletalk Bangladesh Ltd. (Teletalk)	4.254
Total	**130.881**

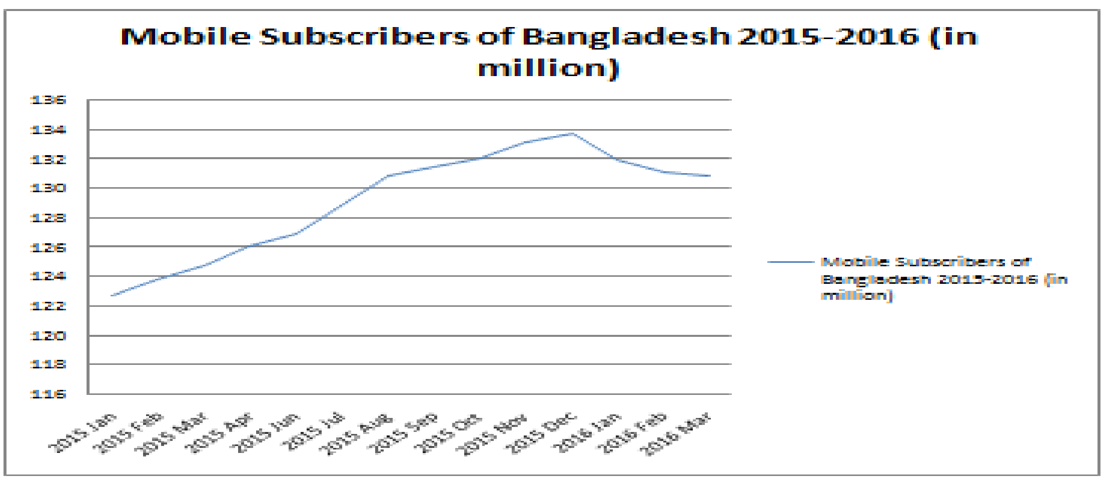


Fig 1: Mobile Subscribers Growth in Bangladesh

In the further chapters we will discuss briefly about the specifications of **QoS** parameters

Chapter 2

Specifications

These specifications include the followings:

a)Proper network planning with the provision of an effective further expansion:

The radio network planning process can be divided into different phases. At the beginning is the Preplanning phase. In this phase, the basic general properties of the future network are investigated, for example, what kind of mobile services will be offered by the network, what kind of requirements the different services impose on the network, the basic network configuration parameters and so on. The second phase is the main phase.

A site survey is done about the to-be-covered area, and the possible sites to set up the base stations are investigated. All the data related to the geographical properties and the estimated traffic volumes at different points of the area will be incorporated into a digital map, which consists of different pixels, each of which records all the information about this point. Based on the propagation model, the link budget is calculated, which will help to define the cell range and coverage threshold. There are some important parameters which greatly influence the link budget, for example, the sensitivity and antenna gain of the mobile equipment and the base station, the cable loss, the fade margin etc.

Based on the digital map and the link budget, computer simulations will evaluate the different possibilities to build up the radio network part by using some optimization algorithms. The goal is to achieve as much coverage as possible with the optimal capacity, while reducing the costs also as much as possible. The coverage and the capacity planning are of essential importance in the whole radio network planning. The coverage planning determines the service range, and the capacity planning determines the number of to-be-used base stations and their respective capacities. In the third phase, constant adjustment will be made to improve the network planning. Through driving tests the simulated results will be examined and refined until the best compromise between all of the facts is achieved. Then the final radio plan is ready to be deployed in the area to be covered and served. The whole process is illustrated as the figure below:

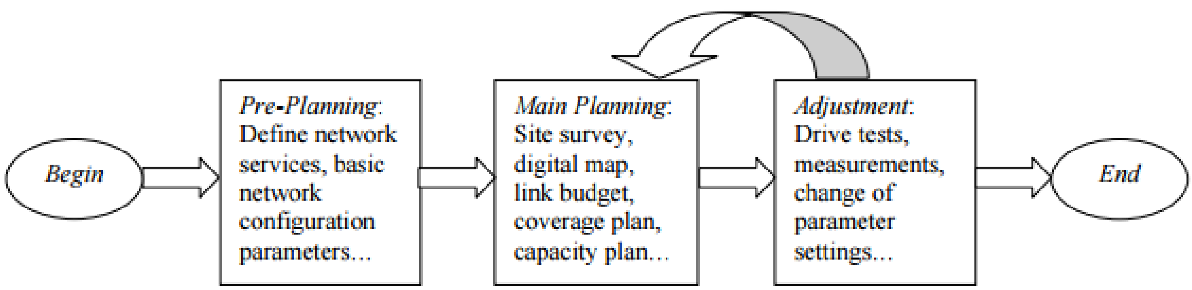


Fig.1 Radio network planning process

In GSM, the network is divided into a lot of cells, and usually a base station is planted in the center of each cell. For the sake of easy analysis, the cells are represented as neighboring hexagons, while in reality they can be of any kind of forms and overlap with each other. The size of each cell, when fixed, will usually stay stable. There is one important feature in GSM network planning: the coverage planning and capacity planning are independent. The coverage planning depends on the received signal strength, that is to say, the covered area is nearly only limited by the minimum signal strength at the cell range, while the later capacity planning depends mainly on the frequency allocation. The link budget is the table recording the power loss in the uplink or downlink of the network. Below is an example of the link budget from GSM 900 MHz.
The link budget results can be improved by adopting some techniques like frequency hopping or using receiver diversity.

In GSM 900 system, there are 125 channels in both uplink and downlink, and these channels span the available bandwidth of GSM 900. The frequency is a scare resource in GSM system, and the frequency must be carefully planned to be reused. The frequency reuse factor is defined as the number of base stations that can be implemented between the current base station and the ones before the same frequency is reused. The antenna height can also influence the reuse factor, since the higher the antenna is, the greater the possibility that the signal causes more interference. Frequency planning is done using one of the previously mentioned optimization algorithms, by setting an adequate cost function to maximize the capacity of the network while minimizing the number of frequency sub band iss used.

b) <u>Stability and reliability of the network</u>:

The number of applications available for wireless communications is growing rapidly: mobile telephony is ubiquitous nowadays, wireless hotspots are spreading everywhere, and also ad hoc networking is growing mature these days. A key characteristic of these scenarios is the dynamic behaviour of the involved communication partners. Communication protocols will have to deal with a frequently changing network topology.

However, many applications require stable connections to guarantee a certain degree of QoS. In access networks, access point handovers may disrupt the data transfer. In addition, service contexts may need to be transferred to the new access points, introducing additional overhead and delays to the connection. In ad hoc networks, mobile services enable peer-to-peer connections for voice or data traffic. Using stable links is crucial for establishing stable paths between connection peers. Rerouting is especially costly in these networks without infrastructure, since it usually results in (at least partly) flooding the network. The stability of a link is given by its probability to persist for a certain time span, which is not necessarily linked with.

This work was supported in part by the German Federal Ministry of Education and Research (BMBF) as part of the IPonAir project (http://www.IPonAir.de/). its probability to reach a very high age. Little work has been published so far on this topic. The related concept of signal stability, well known from cellular networks, has been used to find the right time for a handover. Variations in the received signal strength may hint to the movement pattern of the connection peers and thus allow an estimation of a probable connection loss. However, received signal strength is largely dependent on actual radio conditions. Due to fading effects those measurements are subject to large fluctuations. Another method of estimating the distance and the relative speed of mobile nodes is the use of GPS receivers. But generally, using GPS is infeasible under many conditions. It is e.g. unavailable in indoor environments and it is not suitable for small devices due to its high power consumption. Using the current age of a link to predict the residual lifetime is a relatively unexplored approach, and fundamental research on the relation of a link’s age to its residual lifetime is still missing. This raises the question if a simple and reliable method for stable link selection may be based upon a statistical analysis of link durations.

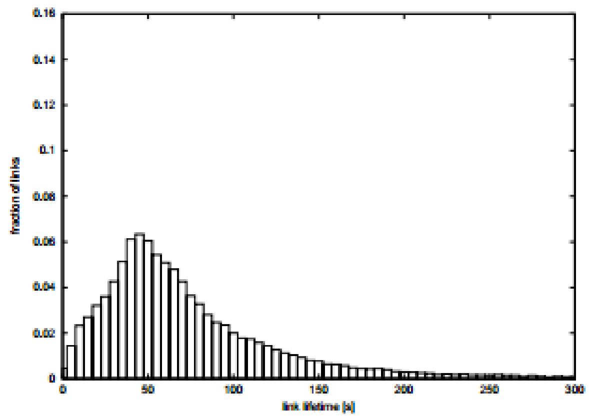


Figure 1. Distribution of link durations in a Gauss-Markov scenario

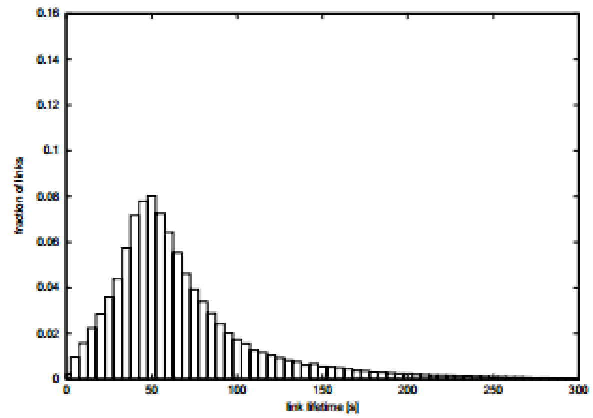


Figure 3. Distribution of link durations in a Random Waypoint scenario

Reliability is a complex notion, which relates to several stability and performance related metrics. Here, we propose a model where the reliability of a network is measured at different levels, reflecting increasing value for the user. A high level picture of the framework is shown in Fig. 1. The proposed model is a generic framework for describing the experienced reliability in MBB networks. In this work, we select a few relevant metrics at each level, and use these to characterize reliability of the measured networks. Other metrics can later be added to give an even more complete picture.

c) Periodic monitoring and proper optimization of different network performance parameters quickly analyze fault, availability, and network performance issues

- Reduce network downtime
- Automatically detect and alert on network performance issues before users start to complain.

Quickly resolve network connectivity issues

Network troubleshooting for tricky intermittent issues with historical performance tracking.

Accelerate time-to-resolution

Identify the root cause faster by isolating the problem to the application or the network.

Wireless Network Optimization

Development Trend

With the development of network services, richness of service types, and diversification of terminals, market competition intensifies. Development of wireless network optimization shows the following trends:

- Wireless network optimization is developed towards the intelligent and intensive direction.
- Data mining is performed for large amount of data collected by network devices are unlocked, and network parameters and settings can be adjusted intelligently and automatically. Tools are applied to improve work efficiency, reduce optimization and maintenance costs, achieving intensive optimization management.
- The development of data services arouses great needs for bandwidth, running short of frequency spectrum resources. Therefore, wireless network optimization focuses on improving the utilization efficiency of frequency spectrum resources and adjusting the resources based on service requirements.
- Market competition intensifies, so user experience becomes a focus. Network optimization turns to service-level and user-level optimization from equipment-oriented optimization.
- Networks for different systems will coexist for a long time. Therefore, in the future, one important task of network optimization is to bear services properly in the networks and to coordinate the networks.

Challenges to Customers

- Market competition intensifies, and new services are developed constantly, so operators focus on how to keep and improve user experience.
- Rapid development of data services brings great pressure to networks. Operators need to provide support for service development through the existing networks, and construct new networks to adapt to service development.
- Solutions are required in special scenarios, such as high-speed rail, campus, and ocean.
- After networks have already operated for several years, a large number of problems emerge. It is necessary to discover and solve basic interference and antenna feeder problems quickly and effectively.
- Full use of tools is required to improve the work efficiency of network operation and optimization.
- When networks for different systems coexist, it is necessary to coordinate operation and optimization of the networks.

ZTEs Solution

- **Routine Network Optimization**

Routine optimization of wireless networks includes: network performance indicator monitoring, troubleshooting, top cell handling, periodic tests and analysis, network access optimization for newly-constructed sites, handover optimization, coverage optimization, frequency and scrambling code optimization, 2G/3G interoperation, and customer complaint handling. Routine network optimization guarantees stable network operation and KPIs, uplifts network performance, and promotes favorable user experience.

- **Voice Quality Improvement**

Voice service is a basic service of carriers. High voice quality is an edged tool for competition. Voice quality improvement of ZTE involves in voice service estimate, voice service troubleshooting, and new voice function applications. The voice service related functions provided by the ZTE's NetMAX optimization tool improve work efficiency and rapidly promote the voice service quality

.

- **Data-Service Quality Improvement**

With popularization of smart terminals and explosive development of mobile Internet services, traffic proliferates. How to guarantee users' data-service experience becomes a problem of operators. ZTE provides the data-service improvement service to solve this problem. Based on basic optimization analysis of networks, and special optimization for different types of services, with newly-developed special features, data-service quality improvement guarantees support for data service development, and provides favorable user experience.

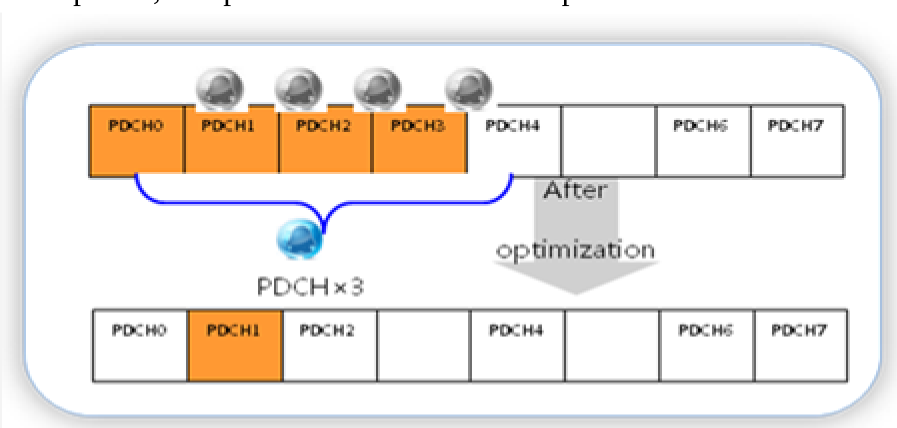


Fig: Optimization Management

- **Capacity Management**

In networks, traffic is unbalanced and tidal. To ensure that users can use services properly, operators implement service development based on maximum configurations. This brings unbalanced network utilization and tidal phenomena. How to solve this problem is a difficulty that operators face to.

Rapid development of data services and explosive increment of traffic speeds up exhaustion of wireless network resources. Operators need to consider how to use existing resources effectively and how to construct networks in the future.

The capacity management service provided by ZTE starts from network resource monitoring, resource adjustment between busy and idle periods, network resource adjustment for tidal traffic, and future network development. The capacity management service solves the insufficient-resource problem by properly adjusting network resources such as frequencies, code channels, devices, and boards. When requirements cannot be met after resources are adjusted, the capacity management service helps operators to perform elaborate capacity expansion planning, PCC/QoS management and control, and WLAN traffic shunting. In this way, operators can take specific measures to gain quality of experience with the least investment to the maximum extent.

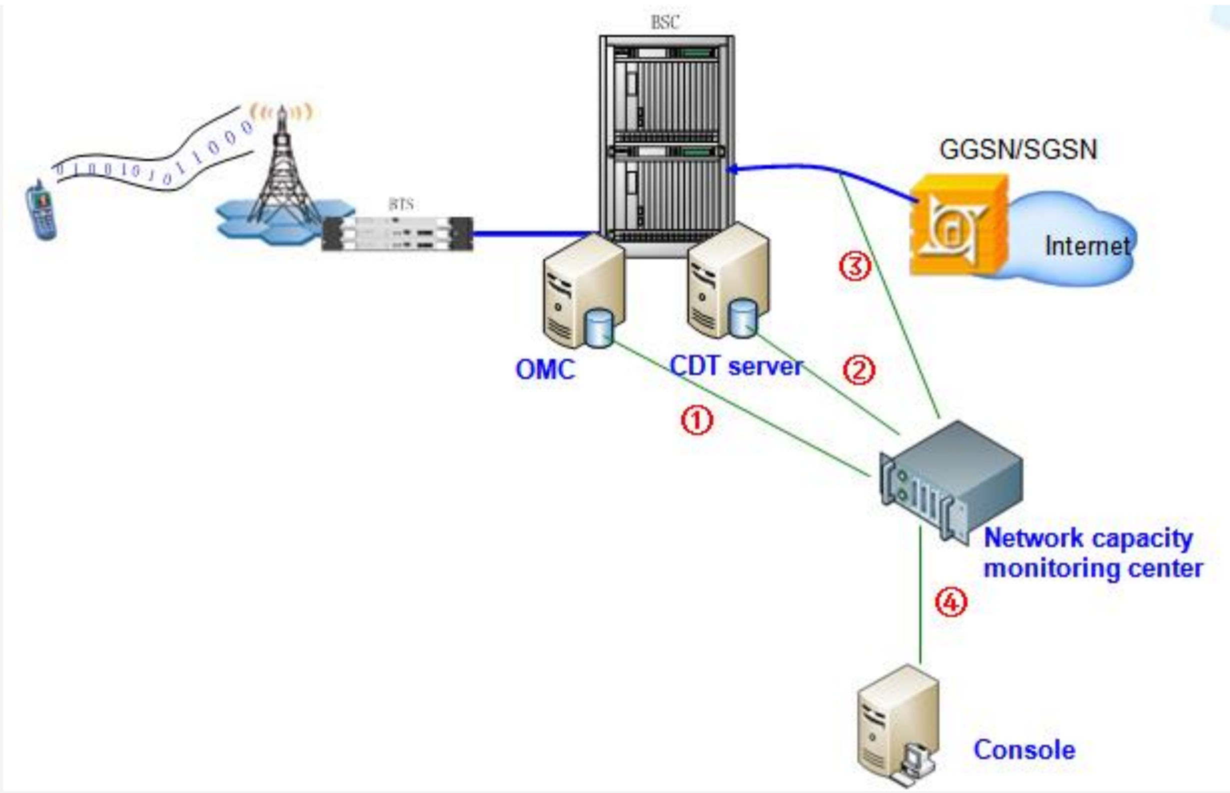


Fig: Capacity Management

User requirements are becoming higher and higher. Primary equipment functions and features are not competent in supporting new services. As a result, operators cannot achieve some new thoughts for improving user experience and system utilization, saving energy, and reducing cost by optimizing the existing system functions. Instead, operators need to develop new functions based on user requirements.

ZTE provides proper design solutions based on user requirements and existing networks. Through high-efficiency development and strict tests performed by professional R&D teams, ZTE can help carriers to implement services and perform real-time monitoring and further optimization.

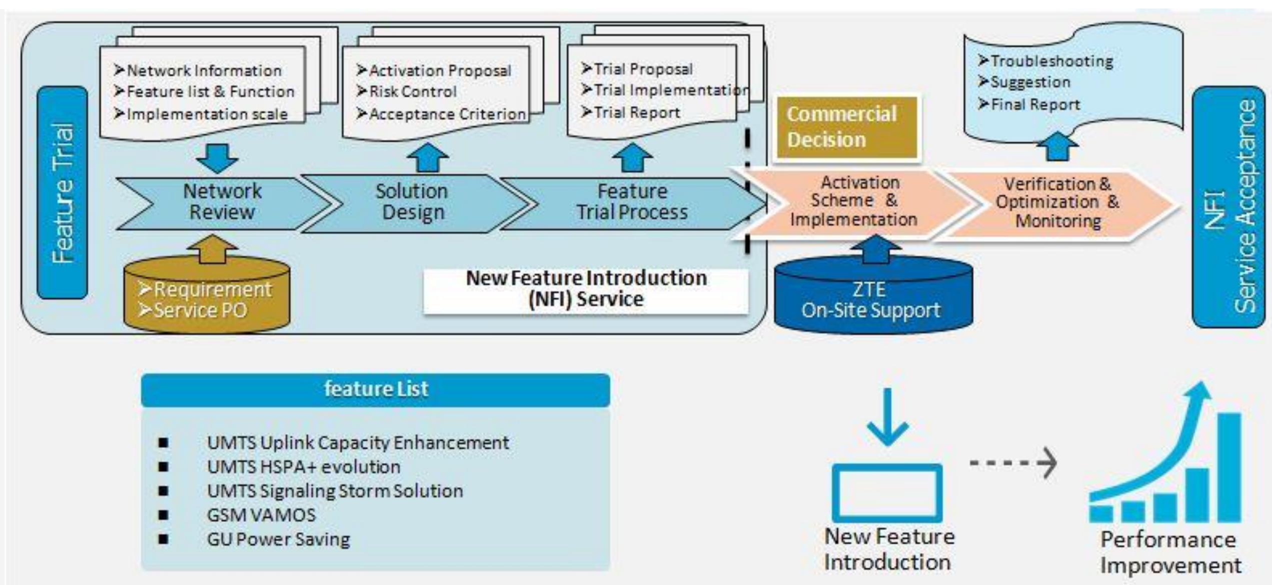


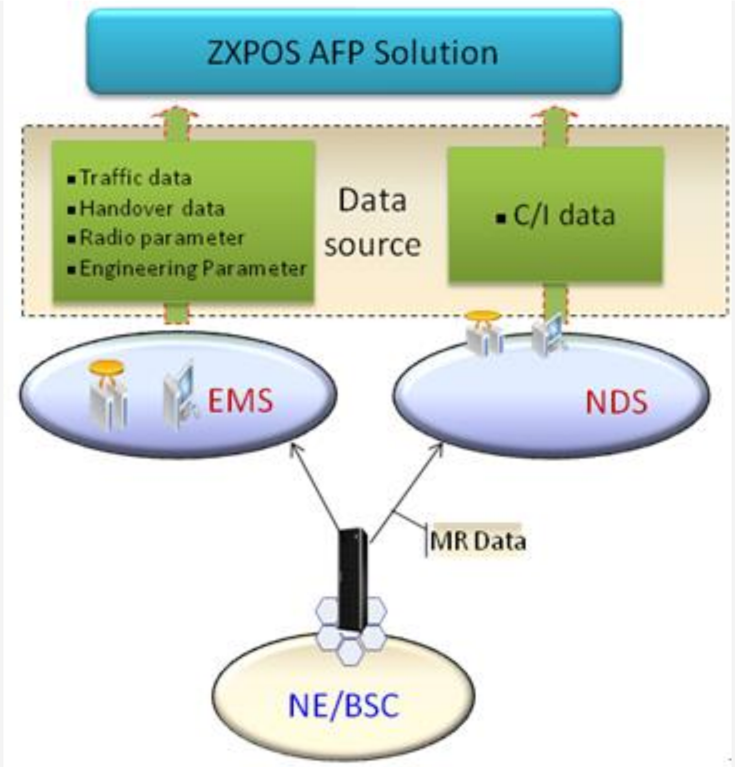
Fig: Important Functions and Features

Intelligent Antenna Feeder Detection

For antenna feeder system problems such as performance degradation, intermodulation interference, and incorrect antenna connection during operation and construction, the intelligent antenna feeder detection tool developed by ZTE can rapidly analyze data collected on back-end devices or systems in networks. Together with the high-efficiency antenna troubleshooting algorithm developed by ZTE, the antenna feeder detection tool can quickly locate faulty antenna feeders in networks. This improves fault location and troubleshooting efficiency.

Intelligent Parameter Planning Solution

In a wireless network, improper configuration for frequencies, code resources, and neighbor cells results in interference and call dropping. Before networks are constructed, frequencies, scrambling codes, and neighbor cells are planned. With construction of networks and increment of the number of sites, configuration becomes confusing due to optimization and adjustment in networks. This affects network quality. It takes too much time and energy for manual replanning, and there may be errors. ZTE provides intelligent planning solutions for these parameters, such as automatic frequency rearrangement based on MR in GMS networks, automatic frequency and scrambler code rearrangement based on MRR in TDS networks, and intelligent PN replanning and neighbor cell reconstruction in CDMA networks. These services can quickly and effectively replan important parameters in networks to solve problems caused by frequencies and scrambling codes. This improves network quality.



Benchmarking

Benchmarking analyzes existing networks (including whether there is frequency pollution in networks) and network performance indicators through various methods and tools to retrieve information about networks. Benchmarking also can compare the network of a operator with those of different operators, so that the operator can know advantages and differences between the networks, and then the operator can make specific network development plans. Benchmarking aims at locating differences and seeking improvement methods. The Benchmarking service of ZTE helps operators to develop towards the communications industry benchmark.

d) Addition of different value added services without hindering the network performance

The deployment of new value-added mobile services has had mixed results in terms of adoption rates and revenue generation, despite the fact that mobile operators rely on such services for future growth, in view of saturated markets for the basic voice service.

This paper suggests that innovation and growth in the mobile services industry largely depends on the extent to which structural conditions in the industry support widespread experimentation, collaboration and risk-taking. Based on the results of earlier scenario analysis research on mobile business, the paper explores the potential for growth in value added mobile services under different assumptions regarding competition and collaboration in the mobile industry.

This paper deliberates the strategic challenges and opportunities for mobile operators, service providers and service integrators under four scenarios for the future of the mobile industry. Given the state of competition and collaboration among those three main roles of industry players in different scenarios, the paper examines the potential for growth in value added mobile services.

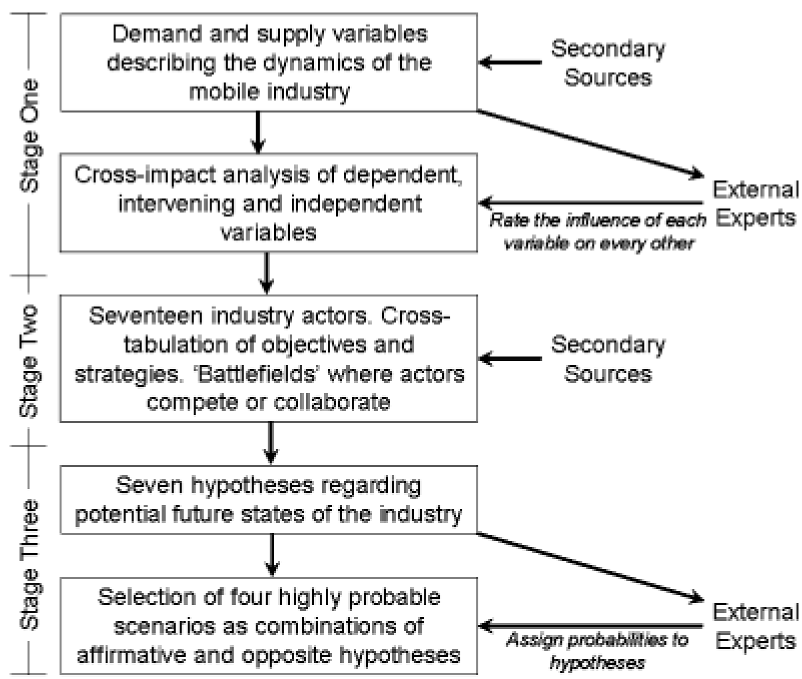


Figure 1. The scenario derivation process

Chapter 3

Network performance:

Network Performance is the most important QoS parameter for the measurement of quality of a Telecom Operator. In order to ensure customer satisfaction different performance parameters needed to be maintained with highest quality. The Network Performance parameter can be categorized as a Key performance Indicator followings:

Chapter 3.1

a)Call Success rate:-

- Represents {No. Successful Calls, Total Call Attempts } as an ordered pair
- Motivation drawn from ASR (Answer to Seizure Ratio) used in PSTN. Difference is, definition of call success determined based on Reason for Call Disconnect (Disconnect Cause code). Ex: A call that was alerted but did not connect is counted as successful. Similar for a call that got disconnected because the called party is busy.

Reported as an exponential moving average over previous time quantum's

The successful call set up consists of two procedures. The simplified description of these procedures is provided in the next text in such a way that the focus is only on the parts necessary to understand the philosophy of Call Set up Success Rate calculation correctly.

First procedure is Immediate Assignment procedure which is used to create a signaling connection between the Mobile station (MS) and the network. It can be initiated only by the MS sending a CHANNEL REQUEST message on the Random Access channel (RACH) to the BTS that it requires a signaling channel (SDCCH). This message contains the information field establishment cause and random reference. The establishment cause gives the reason why the MS is requesting a SDCCH.

Possible reasons are: -

Emergency call –

Call re-establishment –

Originating speech call –

Location updating-

Then it comes next signalization between the MS and network in order to activate the signaling channel, recognize the service being requested by the MS, etc. The successful seizure of SDCCH is acknowledged by sending the Establish Indication message from MS to BTS and then to BSC. Further coordination procedure (authentication, ciphering etc.) are now performed on the SDCCH. Second procedure is Assignment procedure which is used to occupy a radio resource (speech channel). The MSC is initiator of this procedure. The MSC sends an ASSIGNMENT REQUEST message to the BSC requesting the assignment of a radio resource (RR). Then it comes next signalization between BTS and BSC in order to allocate and activate a suitable RR (Traffic channel - TCH).If the TCH is successfully seizured by MS, the BSC sends the ASSIGNMENT COMPLETE message. The main reasons for unsuccessful call setups in mobile networks are lack of radio coverage (either in the downlink or the uplink), radio interference between different subscribers, imperfections in the functioning of the network (such as failed call setup redirect procedures), overload of the different elements of the network (such as cells), etc.

The Call Set up Success Rate is one of the most important Key performance Indicators (KPIs) used by all mobile operators. However there is no standard measurement possible for this parameter. Therefore the different operators can measure it differently. How to optimize the BTS coverage area successfully along with better service is the real challenge .In this paper the main motive is to identify the causes of call setup failures in a GSM service test area and necessitate steps to increase the call success rate using RF optimization. RF Optimization is a very important process in any service provider’s operating lifecycle which is a critical set of activities in the life cycle of any GSM wireless network. RF Optimization involves drive testing, post processing, data analysis, recommendations and action steps. Optimization will be continuous and iterative process of improving network quality. By successful optimization, the Quality of Service, reliability and availability of RF Coverage area is highly improved.

VI Call setup failure reasons There could be so many reasons for a poor CSSR. Some are described as follows:

1. Low Signal Strength
2. SDCCH Congestion
3. CM Service Reject
4. TCH Failure Assignment
5. Hardware Problem

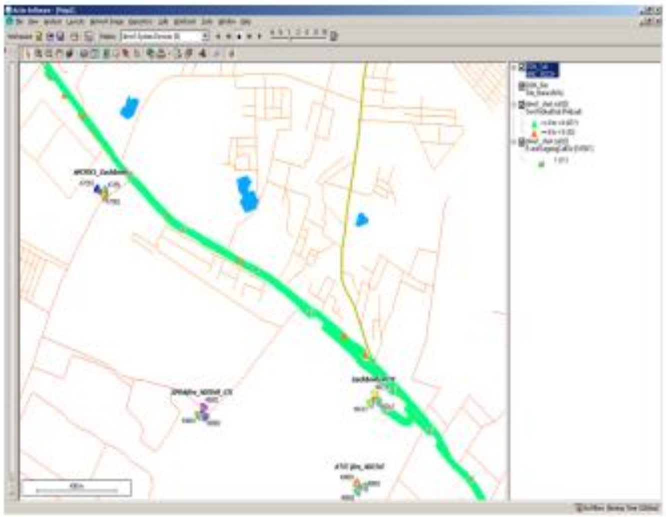
Optimization solutions:

1.Low Signal Strength

Low signal strength call setup failure causes due to weak signal, handover has not taken so that re selection is not happen. Solution By boosting up the signal strength, Rx level will increase due to this better handover taken place And re selection is happened .

2.SDCCH Congestion

No Access to SDCCH. BSS detects channel request (in the form of RACH) from a source, requesting resources for networks transactions. After validation of the RACH, BSS will attempt to allocate a dedicated channel (SDCCH) for the source. One the availability of SDCCH channel is confirmed, the BSS will send immediate assignment to MS indicating the dedicated SDCCH sub-channel (via AGCH), where by subsequent message exchange will be preformed over the dedicated SDCCH.



Short Call – Call setup failure

<u>Case a</u>: Valid RACH (SDCCH Congestion) Due to unavailability of SDCCH, BSS will response to MS with immediate assignment reject, terminating the transactions. In which case, call setup is termed as unsuccessful due to SDCCH congestion. Invalid RACH (Invalid established cause detected in the received RACH)

<u>Case b</u>: Phantom RACHs The received RACH is in fact generated from an "unknown source", where by it fails to continue the transaction after SDCCH has been allocated by the BSS. For instances, case of channel request detected by overshooting cells, handover access burst from distanced MS, hardware deficiency, UL/DL imbalance path, MS moving out of range would carry the Phantom RACHs symptoms. Solution Within the optima there are certain states which can be monitored before coming to conclusion that there is SDDCH problem: a. SDCCH Blocking b. SDDCH Congestion (Valid RACH) If the SDCCH blocking greater than 1% or SDCCH Congestion greater than 2% than that mean that it is a capacity related issue and more slots should be assigned for SDCCH. A TCH can be allocated by passing SDCCH. A parameter namely Immediate Assign Mode when enabled allocates TCH by passing SDCCH.

<u>3.CM Service Reject</u>

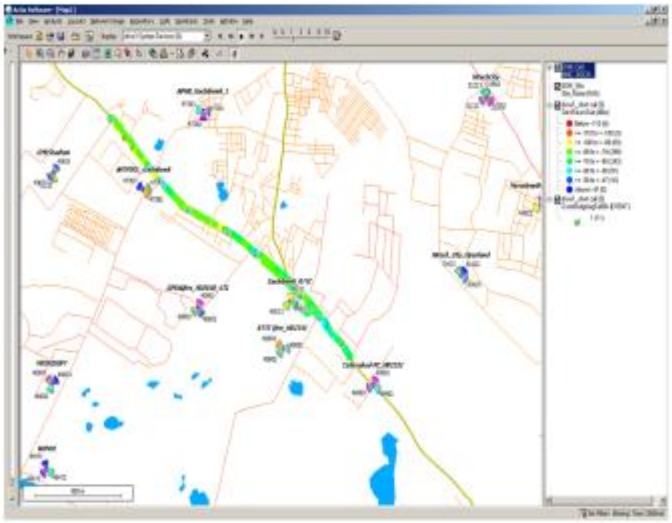
CM Service Request (MOC) or Paging Response (MTC) to BSC/MSC. Inside the CM service request message (MS initiated service request), MS informs the network the types of service it requires, whereby paging response is specific to MTC. Subsequently, BSS embraces the information with its own initiated connection request BSSMAP message, send to MSC to approval. MSC will response with either connection confirmed, confirming the success in link establishment between MS-BSC-MSC, and connection Refused, Indicating the termination of the specific network transaction.

4. TCH Failure Assignment

Upon completion of MS/BSC/MSC link established, MS issues Assignment Request to BSC, Requesting TCH Assignment to the dedicated MS. Subsequently, BSS will attempt to allocate free TCH for MS voice messaging. Once Assignment Command is received by MS, stating the availability of TCH for the MS, it will move to the dedicated TCH and responds with Assignment Complete. In turns, BSS will submit Assignment Complete to MSC as to complete the signal activity. Case TCH Congestion Solution For TCH Congestion certain features can be enabled like TCH queuing, Directed Retry and Congestion Relief. In case of the TCH queuing feature is enable, MS will queue in the Original SDCCH, waiting for the next available TCH. It is to be reminded that once Queuing timer expires. BSS will also terminate transactions, in which case, call setup is termed as unsuccessful due to TCH Congestion. The same situation also applies in situation where Congestion Relief feature is enabled. In the case of Directed Retry feature is enabled, MS will perform Handover to TCH of another cell if a valid handover neighbor is detected. The best thing to do is to add more radios in the cell to remove congestion. Interference analysis on a particular carrier can be done through an optimization tools like Neptune. Once interfering frequencies are determine, the frequency plan can be cleaned from such frequencies.

5. Hardware Problem

Hardware failures also play major role for poor CSSR. Improper functionality of any BTS hardware can affect the overall performance of sites. Solution: If there are no capacity or RF issues then equipment needs to be checked. Before starting the drive test make sure that the cell site are free for any hardware alarms. The important parameter to check is the path balance. If path balances are not fine then start checking the power from radio to connected antennas. If we take the examples of GSM 900 scenario, the link budget defines that the radio should transmit 40 watts power and at the top of the cabinet, 20 watts are received. While checking the power, if any components seem to procedure more losses than expected, change that component. Similarly check the power at antenna feeder ports. Some time due to the water ingress, connectors get rusty and needs to be replaced.



Short Call- Signal Quality

b. Call set-up time

In telecommunication, the term call set-up time has the following meanings:

1) The overall length of time required to establish a circuit-switched call between users.

2) For data communication, the overall length of time required to establish a circuit-switched call between terminals; i.e., the time from the initiation of a call request to the beginning of the call message.

Note: Call set-up time is the summation of: (a) call request time—the time from initiation of a calling signal to the delivery to the caller of a proceed-to-select signal;

(b) Selection time—the time from the delivery of the proceed-to-select signal until all the selection signals have been transmitted; and

(c) Post selection time—the time from the end of the transmission of the selection signals until the delivery of the call-connected signal to the originating terminal.

Under the general heading of quality of experience (QoE) one of the more noticeable points faced by the user is the apparent delay in voice call set up time. The call set up delay can be defined as the time interval from the instant the user initiate a connection request until the complete message indicating call disposition is received by the calling terminal. When establishing a connection the user, due to this delay, may think that the call has not gone through or the network is not responding which may prompt the user to re-dial or in some cases to abandon the connection attempt. Users can experience similar delays during the establishment of packet-based services such as Internet browsing. From the service provider's perspective maintaining an acceptable quality of service is very important. This issue decreases the users perception of the network performance and efficiency. This work item is intended to investigate mechanisms to improve the connection establishment times and implement those changes in the specifications.

The delay in call set up can be attributed to:

- Processing time in the UTRAN
- Processing time in the Core network
- Processing time in UE
- Call setup and alerting phase in the core network
- UTRAN and CN Protocols and associated overhead including protocol conversion

- Signaling delay on the air interface
- Signaling delay on UTRAN interfaces and towards CN
- NAS procedures

c. Call drop rate

The objective of this parameter is to provide the consumer with an expectation of how successful a mobile network will be at retaining the signal throughout the whole duration of the call. ETSI EG 202 057-3 v1.1.1 (2005-04) defined Dropped Call Ratio as “The percentage of calls which, once they have been correctly established and therefore have an assigned TCH, are interrupted prior to their normal completion by the user, the cause of the early termination being within the operator’s network”. 2.3. The formula for calculating the percentage of dropped calls is:

s = □ □ x 100 where: A = the total number of interrupted calls (dropped calls) B = the total number of calls successfully established (where traffic channel is allotted)

The formula includes the interrupted calls which consist of failures which cause the dropping of the call once the TCH has been successfully established, and the successful seizure of TCH for an originated or terminated call. The total number of established calls shall include the number of TCH assignment in a cell for establishment of new call + number of TCH assigned for incoming handover – number of TCH made free for outgoing handovers.

Table 2.1: TCH call drop rate

CH drop (in %)	COUNT OF CELLS (TSP 1)	COUNT OF CELLS (TSP 2)	COUNT OF CELLS (TSP 3)
2-4%	5	10	473
4-5%	1	0	15
5-10%	0	0	12
>10%	0	0	4
Count of cells not meeting TRAI benchmark	6 out of 8604	10 out of 15850	504 out of 16117

Table 2.2 Call drop counters

Counter name	Reason for call drop
TSUDLOS	Dropped calls due to Sudden Loss
TDISSUL	Dropped calls due to insufficient signal strength on the Uplink
TDISSDL	Dropped calls due to insufficient signal strength on the Downlink
TDISSBL	Dropped calls due to insufficient signal strength on Both link
TDISQAUL	Dropped calls due to Bad quality on the Uplink
TDISQADL	Dropped calls due to Bad quality on the Downlink
TDISQABL	Dropped calls due to Bad quality on Both link
TDISTA	Dropped calls due to excessive Timing Advance

Worst affected cells having more than 3% TCH drops: As discussed earlier, the TCH call drop does not reveal the extent of number of areas or localities with worst call drop rate. Worst affected cells are defined as cells in which the call drop rate exceeds 3% during cell Bouncing Busy Hour (CBBH) or at any other hour of a day. An analysis of this parameter was carried out for various TSPs in Delhi Circle for the month of August 2015 and the results tabulated at Table 2.3 below. Though the operators are generally meeting the TRAI bench mark of <=3%, a drill down of the values at cell level shows that for one TSP, there are more than 300 cells not meeting the benchmark of 3%.

Worst affected cells having more than 3% TCH drops (Call Drop Rate)	Count of cells (TSP 1)	Count of cells (TSP 2)	Count of cells (TSP 3)
	108 out of 8604	106 out of 15850	335 out of 16117

Connections with good Voice Quality

An analysis of this parameter was carried out for various TSPs in Delhi Circle for the month of August 2015 and the results tabulated at Table 2.4 below. Though the operators are generally meeting the bench mark of >95%, a drill down of the values at cell level shows that there are around 15 % cells not meeting the benchmark.

CONNECTION QUALITY (in %)	COUNT OF CELLS (TSP 1)	COUNT OF CELLS (TSP 2)	COUNT OF CELLS (TSP 3)
90-95%	701	2473	71
85-90%	91	456	8
80-85%	14	130	3
70-80%	6	83	0
50-70%	3	19	2
40-50%	0	4	0
20-40%	0	3	0
5-20%	0	1	0
<5%	0	0	0
Count of cells violating TRAI benchmark	815 OUT OF 7545 CELLS	3169 OUT OF 15348 CELLS	84 OUT OF16108 CELLS

Customer dissatisfaction arises because of calls getting dropped after establishment. But there could also be customer dissatisfaction, because of failure during establishment of a call. To analyses these failures, the various cause codes for failures during the process of establishment of a call, was obtained from TSPs for the month of August 2015. These cause codes are recorded by the system, which provide an insight into the major causes due to which a call was unsuccessful. The various cause codes collected from TSPs were analysed. Currently there are no regulations concerning these cause codes. However, 'The Standards Of Quality Of Service Of Basic Telephone Service (Wire line) And Cellular Mobile Telephone Service Regulations, 2009' for Basic (Wire line) Services have defined the benchmark for Call completion Rate3 (CCR) within a local network at ≥ 55% or the Answer to Seizure ratio (ASR) at≥ 75%.

Failure during call establishment could also be due to congestion in the signalling channel known as SDCCH (in respect of GSM network) /Paging Channel Congestion (in respect of CDMA network) or in the TCH. SDCCH channel/paging channel is the control channel where majority of the call set 0 10 20 30 40 50 NORMAL END OF THE CALL B SUBSCRIBER BUSY ABSENT SUBSCRIBER B ANSWER TIME OUT CIRCUIT RELEASED BY CO-EXCHANGE NO RESPONSE TO CALL… NETWORK, UNALLOCATED… CLEAR/A ONHOOK DURING WAIT… CALL REJECTED Others TSP 1 TSP 2 TSP 3 TSP 4 0 50 NORMAL END OF THE CALL CLEAR/A ONHOOK DURING WAIT FOR… B SUBSCRIBER BUSY B ANSWER TIME OUT CLEAR/A ONHOOK DURING SET-UP PHASE ABSENT SUBSCRIBER NO RESPONSE TO CALL ESTABLISHMENT;… NETWORK, UNALLOCATED NUMBER CALL REJECTED OUT OF RADIO COVER, RE-ESTABLISH FAIL

Others TSP 1 TSP 2 TSP 3 TSP 4 17 up occurs and is used for mobile station (mobile handset) to Base Transceiver Station (BTS) communications before the mobile station is assigned TCH/speech channel. TCH is a logical channel which carries either encoded speech or user data. If there is no free channel in radio access network to establish a call, it will lead to blocked call. Hence, connection establishment (accessibility) represents congestion in the radio access network.

Short Duration Calls: Another variant to analyse call drops could be an analysis of the Call Data Records in the billing system. It may be noted that short duration calls do not necessarily mean a call drop. The Call Data Record analysis in Delhi was made for some service providers for the month of August 2015. This could imply either the calls were made for short duration or the calls were dropped within 30 seconds. 2.16. It was also noticed that some of these calls are also repeat calls which might indicate multiple failures in getting connected to the same number. The repeat calls which last for less than 60 seconds as a percentage of total calls (i.e. of all possible durations) has been shown in figure 2.9. While this percentage is small, this might contain calls dropped. The data varies with the various TSPs, with an average of around 5%. The differences in prepaid and postpaid call drops are negligible in this regard.

FACTORS RESPONSIBLE FOR CALL DROP

3.3.1. Due to the increase in users' demand for wireless cellular connectivity, and to accommodate more number of users, the cell size in mobile wireless cellular networks is getting reduced specially in urban areas. Due to this, more number of handovers (or handoffs) are taking place and the probability of calls dropping has increased. Also, if sufficient bandwidth is not available for new calls, then blocking probability for newly generated calls also increases. Dropping of calls during handover is less desirable than blocking of new calls.

3.3.2. Another reason for dropped calls is when a mobile user enters an area without adequate signal strength, or the signal has been interrupted, interfered with, or jammed. From the network perspective, this is similar to leaving the coverage area. Occasionally, calls are dropped upon handoff – between cells. One possible reason for such occurrences is traffic imbalance between two cell sites when the new cell site is at capacity and cannot accept additional traffic from the cell trying to "hand in".

3.3.3 In networks there could be sites added or modified or moved even on a daily basis. This requires constant updating or reconfiguration of the networks. Wrong network configuration or wrong neighbor definition can be another reason, which renders one cell site "unaware" of the cell the mobile user is attempting to hand off to. If the mobile phone cannot find an alternative cell to provide the handshake, then the call is lost.

3.3.4 In cellular networks, co-channel and adjacent channel frequency interference is mainly caused by neighboring cells Thus cells in the coverage area of the serving cell should be checked. The density of the base station greatly varies in different regions. The distances between base stations are much longer in rural areas than in cities. Thus, the cell coverage radius in rural area is much larger than in cities. The cell coverage is then usually indicated by the base station site layer and the azimuth award side (the antenna). Co-channel and adjacent channel interference can also be responsible for dropped calls in a wireless network. Neighboring cells with same frequencies may interfere with each other to deteriorate the quality of service and cause dropped calls. Through drive test the C/I ratio could be checked for levels of the signal strength of the current serving cell to that of the signal strength of undesired (interfering) signal components. The data regarding C/I ratio measured during drives tests conducted in Delhi, Mumbai and other major cities is at Annexure-V. It could be noticed that most of the TSPs in the metros are having relatively higher C/I.

3.3.5 Transmission problems also cause dropped calls due to a faulty transceiver (TRX) within the base station or faulty transmission media. Call drops could also be because of hardware related issues including equipment failure. At the receivers' end, calls may be dropped if a mobile phone loses battery power and abruptly stops transmitting.

3.3.6 The calls could also drop because of various antenna related issues like

- If the transmit antennas of two cells are improperly connected; the uplink signal level in each cell is much lower than the downlink signal level in the cell. Therefore, call drops are likely to occur at places far away from the BTS.
- If a directional cell has main and diversity antennas, the BCCH and SDCCH of the cell may be transmitted from different antennas. If the two antennas have different traffic channel angles or azimuths, the coverage areas of the two antennas will be different. In this case, the mobile station (MS) can receive the BCCH signals from one antenna and when a call is made, the MS cannot seize the SDCCH transmitted by the other antenna and thus a call drop occurs.
- If the feeder cable is damaged, water leaks in the feeder, or the feeder and the connector are not securely connected, both the transmit power and receiver sensitivity of the antenna are reduced. This can also result in call drops.

3. 3.7. In a cellular network, the difference between the uplink signal and the downlink signal level may be high which can further cause call drop, in the following situations: The Transmit power of the BTS is high.

- The tower mounted amplifier (TMA) or BTS amplifier malfunction.
- The antenna and the connector are not properly connected.

3 .3.8. As a result, call drops may occur at the edge of the coverage area Propagation factors on signal behavior such as reflections and multipath, diffraction and shadowing, building and vehicle penetration, propagation of signal over water, propagation of signal over vegetation (foliage loss), fading of the signal, interference could also lead to call failures. Generally more than 50% of the reasons for dropped calls6 in a cell, are reported to be mainly due to electromagnetic causes, as shown at Table: 2.1. In some cases, where the networks use load control algorithm (typically located in the radio network controller), Calls can also be dropped to preserve system quality.

3.3.9. Call could also drop due to irregular user behavior (mobile equipment failure, phones switched off after ringing, subscriber charging capacity exceeded during the call). Other causes can be due to abnormal network response (e.g. radio and signaling protocol error). Reasons of increasing Call Drop Rate in Urban areas and Metros .

3.3.10. The increasing rate of call drops, especially in urban and metro areas, can also be attributed to spectrum related issues.The Authority has recommended to DoT from time to time, for making additional spectrum available in existing as well as new bands for commercial use to serve the ever increasing subscriber base and also to deliver higher data rates.

But it has also been observed that a part of the spectrum remained unsold in the recent auction (22.5 MHz out of 108.75 MHz in 800 MHz band, 9.8 MHz out of 177.8 MHz in 900 MHz, 5.4 MHz out of 99.2 MHz in 1800 MHz and 15 MHz out of 85 MHz in 2100 MHz) due to various reasons.

3. 3.11. The Table 3.2 below, depicts the number of subscribers, the spectrum availability and the number of BTS towers available with some TSPs in Delhi. As can be seen from the table, the spectrum availability has nearly been constant since 2009 for some operators and has decreased for some operators, since they have used a portion of the spectrum for new technology deployments after the auction, despite an increasing number of subscribers. This has led to an increase in the subscriber density per MHz of spectrum. Though the TSPs have installed additional towers, but the growth of BTS sites has not kept up with the growth of the subscriber base. The deficient spectrum availability or non-installation of BTS by the TSPs has translated into declining call quality. Moreover, operators who have started to use a portion of the spectrum obtained in the recent auction for new technologies after renewal of their license, require extensive rebalancing of voice traffic. If this exercise is not carried out properly, it could lead to call drops. prior to the spectrum auction. Non-installation or delayed installation of these filters will result in calls being dropped because of interference.

3.3.12. In metros, some of the operators, who operate in dual-band network, usually setup a call in a GSM1800 cell and hand over the same to a GSM900 cell in the same site. If the network clocks of GSM900 cell and GSM1800 cell are not properly synchronized, it could also lead to call drops, especially in cases of circuit embedded networks.

3.3.13. In some of the major towns, there are objections raised by resident welfare associations for installed mobile towers, because of mounting fears7 about radiation, transmitting from the towers and the perceived health hazards associated with the same. The protests in residential areas have resulted in towers being pulled down or in stalling installation of new towers affecting mobile service quality. Every tower pulled down also exerts additional loads in the neighbouring adjacent cells resulting in poor call quality.

In fact less number of towers in an area will actually increase the power levels of the hand sets, since the mobile handset has to 'shout' so that its signal reaches the BTS. 3.16. Also the users, due to weak signal strength in their building or premise, tend to install signal boosters to boost their received mobile signal strength. More often these users tend to purchase boosters that are not band specific to their service provider and boosts the complete GSM band (including all TSPs), resulting in interference of the signals. In Delhi alone there are more than 250 identified illegal boosters operating in the network.

Chapter 3.3.1

EFFECT OF NETWORK TRAFFIC ON CALL DROP RATE

An attempt has been made in this chapter to analyse the effects of carried traffic intensity and utilization factor of the network on the call drop rate. Average Traffic Intensity

Average traffic intensity is a measure of the average occupancy of a channel during busy hour, measured in traffic units (Erlangs) and is defined as the ratio of the time during which a channel is occupied (continuously or cumulatively) to the time this channel is available for occupancy.

A traffic intensity of one traffic unit (one Erlang) means continuous occupancy of a channel during the time period under consideration, regardless of whether or not information is transmitted8. The traffic intensity offered by each user is equal to call request rate multiplied by the holding time. That is, each user generated a traffic intensity of AU (Erlangs) given by: AU = G X H Where, H=average duration of a call and, G= average number of call requests per unit time for each user. For a system containing 'U' users and an unspecified number of channels, the total offered traffic intensity 'A' is given as: A= U x AU

The average carried traffic of each cell has been directly collected from the TSPs for the month of August 2015. Then the traffic carried by each BSC has been calculated and a correlation between the average carried traffic intensity and call drop rate for various operators has been plotted as shown.

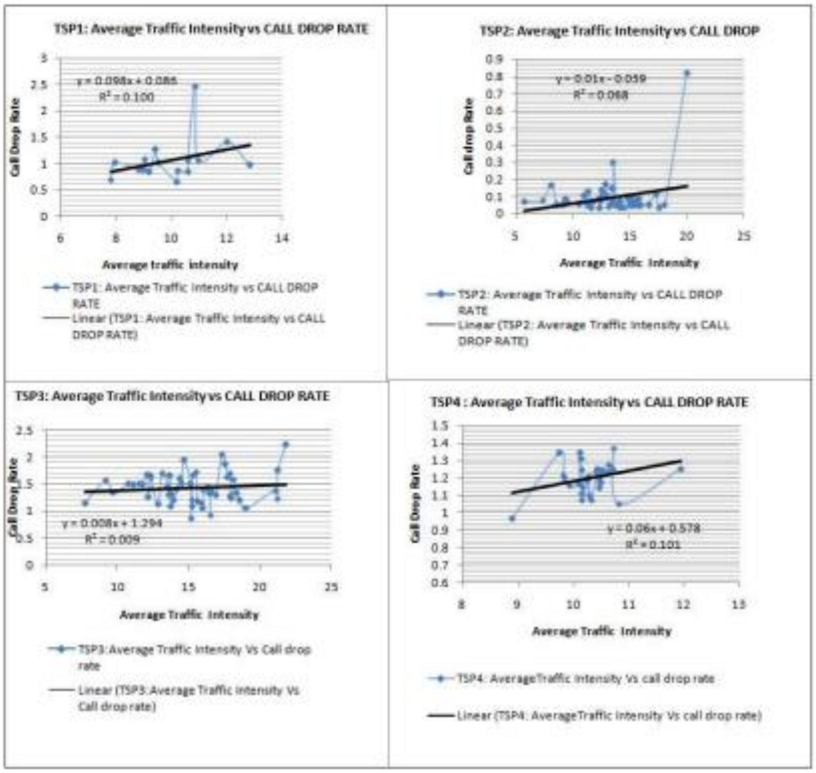


Figure: Average traffic vs call drop

It can be inferred from the above graphs that on an average, as the traffic intensity (on x-axis) increases, the call drop rate increases in a linear fashion. This explains that when the admissible users exceed the maximum threshold (till the point where the call maintenance stage is able to control), the call drop rate increases.

The system performance begins to degrade as the traffic increases beyond the network capacity. If the various radio resources are available during a call maintenance stage, the network becomes stable, thereby 28 providing the required QoS to users accessing the system. This calls for appropriate suitable load control measures to be taken by the telecom service providers during congestion periods.

Chapter 3.3.2

UTILIZATION FACTOR

The utilization factor is the ratio of the time that the network is in use to the total time that it could be in use. Utilization factor is the traffic load in the cellular network. Traffic load signifies the strength of the offered traffic in the network.

By definition, the traffic load is the ratio between the arrival rate of calls and the service rate of the calls arriving. Utilization factor gives the product of total traffic offered and the mean service time. Utilization factor = Traffic Load= Average Traffic Intensity (A) x Mean holding time (H) 4.7. As the traffic load or utilization factor increases, the load on the network increases and hence the probability of call drops also increases.

The correlation between the utilization factor and call drop rate for various operators is shown in the following graphs at Figure 4.2. The utilization factor has been calculated for each day in the month of August 2015 for all the cells of the respective service providers. The mean holding time has been taken as 140 seconds.

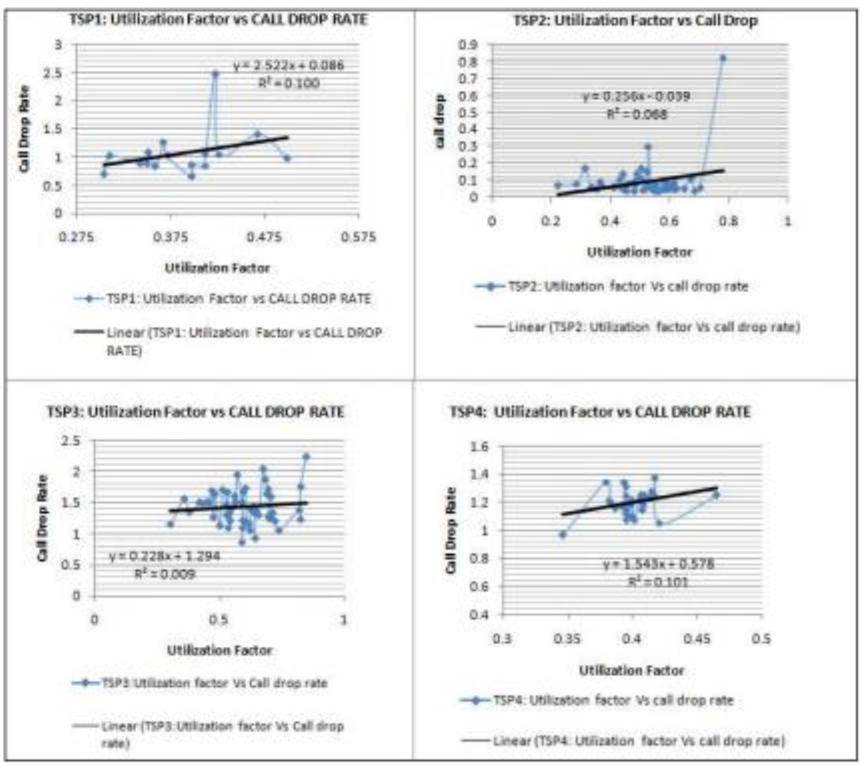


Figure Utilization factor vs call drop

It can be inferred from the above graphs that on an average, as the utilization factor (on x-axis) increases, the call drop rate increases in a linear fashion. From the above analysis, it can be inferred that the Call Drop Rate of a cellular network varies linearly with average traffic intensity and Utilisation factor above a certain threshold in similar fashion.

The tele-traffic modeling of dropped calls performance in an established cellular network can be compared for different service providers to estimate the network capacity problems leading to frequent call drops. Use of directive antennas with realistic pattern can strongly impact the spectral efficiency of the systems and help in reduction of call drops, using the dynamic channel assignment strategy, but this could also load the system. In cellular networks, it is also possible to arrive at the probability of call drops theoretically. The probability that a call is dropped could be calculated based on handoff rate and handoff call arrival rate as given below: Where, pd is the call drop probability; Pf is the Handoff blocking probability or forced termination probability; λ is the new call arrival rate; λh is the handoff traffic arrival rate; and E[H] is the average number of handoffs during the cell life. This indicates that in homogeneous wireless networks, the call dropping probability (which is a network wide quantity) can be completely determined by the call arrival rates for new calls, and the handoff

blocking probability in a single cell. This is consistent with the fact that a homogeneous wireless network can be completely characterized by a single cell in the wireless network.

Chapter 3.4

Introduction to Handover

One of the key elements of a mobile phone or cellular telecommunications system, is that the system is split into many small cells to provide good frequency re-use and coverage. However as the mobile moves out of one cell to another it must be possible to retain the connection. The process by which this occurs is known as **handover or handoff**.

The term handover is more widely used within Europe, whereas handoff tends to be use more in North America. Either way, handover and handoff are the same process.

Requirements for GSM handover

The process of handover or handoff within any cellular system is of great importance. It is a critical process and if performed incorrectly handover can result in the loss of the call. Dropped calls are particularly annoying to users and if the number of dropped calls rises, customer dissatisfaction

increases and they are likely to change to another network. Accordingly GSM handover was an area to which particular attention was paid when developing the standard.

Types of GSM handover

Within the GSM system there are four types of handover that can be performed for GSM only systems:

- **Intra-BTS handover:** This form of GSM handover occurs if it is required to change the frequency or slot being used by a mobile because of interference, or other reasons. In this form of GSM handover, the mobile remains attached to the same base station transceiver, but changes the channel or slot.
- **Inter-BTS Intra BSC handover:** This for of GSM handover or GSM handoff occurs when the mobile moves out of the coverage area of one BTS but into another controlled by the same BSC. In this instance the BSC is able to perform the handover and it assigns a new channel and slot to the mobile, before releasing the old BTS from communicating with the mobile.
- **Inter-BSC handover**: When the mobile moves out of the range of cells controlled by one BSC, a more involved form of handover has to be performed, handing over not only from one BTS to another but one BSC to another. For this the handover is controlled by the MSC.
- **Inter-MSC handover:** This form of handover occurs when changing between networks. The two MSCs involved negotiate to control the handover.

GSM handover process

Although there are several forms of GSM handover as detailed above, as far as the mobile is concerned, they are effectively seen as very similar. There are a number of stages involved in undertaking a GSM handover from one cell or base station to another.

In GSM which uses TDMA techniques the transmitter only transmits for one slot in eight, and

similarly the receiver only receives for one slot in eight. As a result the RF section of the mobile could be idle for 6 slots out of the total eight. This is not the case because during the slots in which it is not communicating with the BTS, it scans the other radio channels looking for beacon frequencies that may be stronger or more suitable. In addition to this, when the mobile communicates with a particular BTS, one of the responses it makes is to send out a list of the radio channels of the beacon frequencies of neighbouring BTSs via the Broadcast Channel (BCCH).

The mobile scans these and reports back the quality of the link to the BTS. In this way the mobile assists in the handover decision and as a result this form of GSM handover is known as

Mobile Assisted Hand Over (MAHO).

The network knows the quality of the link between the mobile and the BTS as well as the strength of local BTSs as reported back by the mobile. It also knows the availability of channels in the nearby cells. As a result it has all the information it needs to be able to make a decision about whether it needs to hand the mobile over from one BTS to another.

If the network decides that it is necessary for the mobile to hand over, it assigns a new channel and time slot to the mobile. It informs the BTS and the mobile of the change. The mobile then retunes during the period it is not transmitting or receiving, i.e. in an idle period.

A key element of the GSM handover is **timing and synchronization.**

There are a number of possible scenarios that may occur dependent upon the level of synchronization.

- **Old and new BTSs synchronized:** In this case the mobile is given details of the new physical channel in the neighboring cell and handed directly over. The mobile may optionally transmit four access bursts. These are shorter than the standard bursts and thereby any effects of poor synchronization do not cause overlap with other bursts. However in this instance where synchronization is already good, these bursts are only used to provide a fine adjustment.
- **Time offset between synchronized old and new BTS**: In some instances there may be a time offset between the old and new BTS. In this case, the time offset is provided so that the mobile can make the adjustment. The GSM handover then takes place as a standard synchronized handover.
- **Non-synchronized handover:** When a non-synchronized cell handover takes place, the mobile transmits 64 access bursts on the new channel. This enables the base station to determine and adjust the timing for the mobile so that it can suitably access the new BTS.
- **Inter-system handover:** With the evolution of standards and the migration of GSM to other 2G technologies including to 3G UMTS / WCDMA as well as HSPA and then LTE, there is the need to handover from one technology to another. Often the 2G GSM coverage will be better then the others and GSM is often used as the fallback. When handovers of this nature are required, it is considerably more complicated than a straightforward only GSM handover because they require two technically very different systems to handle the handover.

These handovers may be called intersystem handovers or inter-RAT handovers as the handover occurs between different radio access technologies.

The most common form of intersystem handover is between GSM and UMTS / WCDMA. Here there are two different types:

- **UMTS / WCDMA to GSM handover:** There are two further divisions of this category of handover:
  - **Blind handover:** This form of handover occurs when the base station hands off the mobile by passing it the details of the new cell to the mobile without linking to it and setting the timing, etc of the mobile for the new cell. In this mode, the network selects what it believes to be the optimum GSM based station. The mobile first locates the broadcast channel of the new cell, gains timing synchronization and then carries out non-synchronized intercell handover.
  - **Compressed mode handover**: using this form of handover the mobile uses the gaps I transmission that occur to analyze the reception of local GSM base stations using the neighbor list to select suitable candidate base stations. Having selected a suitable base station the handover takes place, again without any time synchronization having occurred.

Handover from GSM to UMTS / WCDMA: This form of handover is supported within GSM and a "neighbor list" was established to enable this occur easily. As the GSM / 2G network is normally more extensive than the 3G network, this type of handover does not normally occur when the mobile leaves a coverage area and must quickly find a new base station to maintain contact. The handover from GSM to UMTS occurs to provide an improvement in performance and can normally take place only when the conditions are right. The neighbor list will inform the mobile when this may happen.

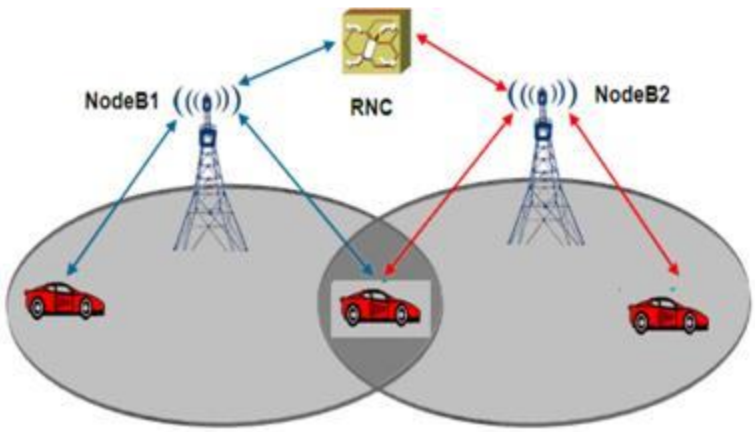


Fig: Handover among the BTS

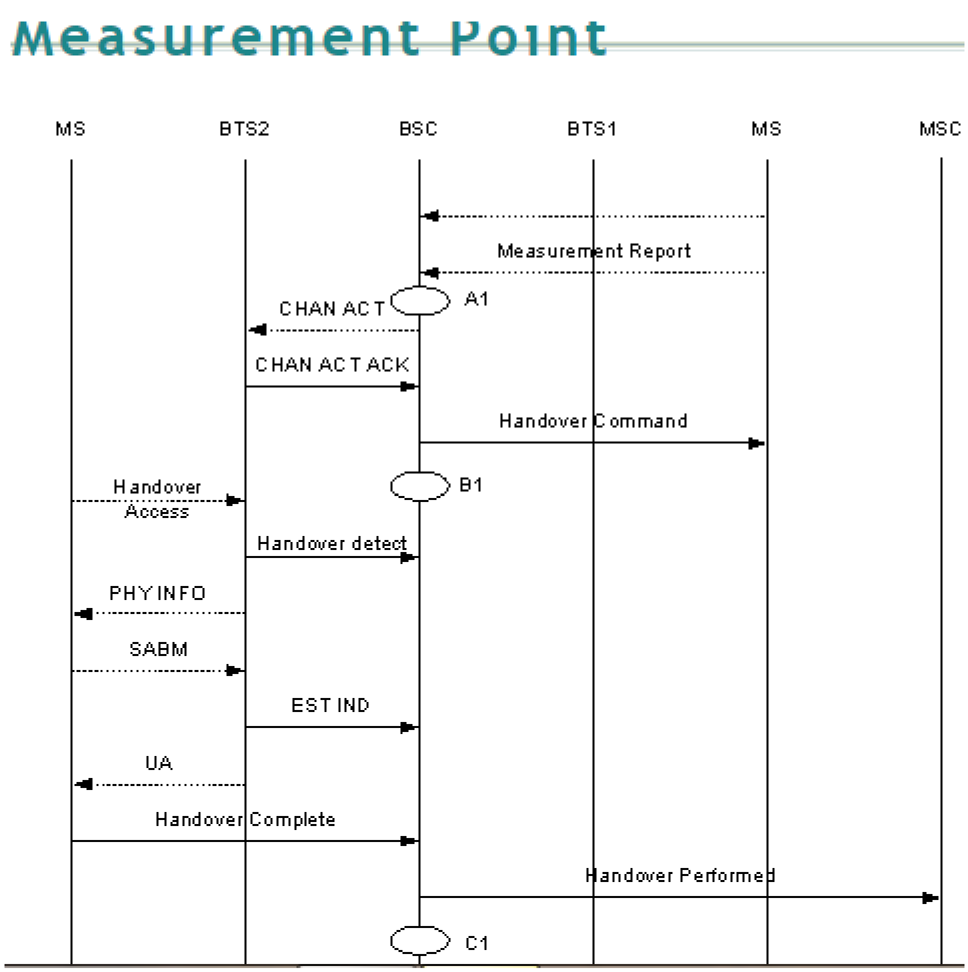
d. Handover Success Rate:

The HOSR is an important KPI of the call hold type. According to the processes, this KPI can be divided into two types: Handover Success Rate and Radio Handover Success Rate. According to the relations between involved network elements (NEs), this KPI can be divided into three types: Success Rate of Intra-BSC Handover, Success Rate of Incoming BSC Handover, and Success Rate of Outgoing BSC Handover. The HOSR is an important KPI assessed by operators because the value of the HOSR directly affects the user experience.

The HOSR is obtained through traffic measurement. The recommended formula for calculating this KPI is as follows:

- **Handover Success Rate = Successful Handovers/Handover Requests**

- **Radio Handover Success Rate = Successful Handovers/Handover Commands**



Handover in GSM is very important to keep continuous of call communication

Handover also purposed to improve services performances and reduce congestion in the cells.
Some other types of Handover system
Based on reason type of handover in GSM can be classified as below
1.Emergency Handover : Timing Advance(TA) Handover, Bad Quality Handover ,RxLevel Drop handover and Interference Handover.
2. Load Handover
3. Normal Handover

4.Edge Handover

5.Layer Cell Handover and

6.Power Budget(PBGT) handover

s7. Fast Moving Handover

8. Overlaid/Underlaid Handover

If HOSR will be good TCH drop will also be good.
If Handover success rate degrades call drop rate will take place.

Reasons for Poor HOSR:

- Improper Neighbor planning.
- CO-BCCH-BSIC issues in Neigh.
- Parameter Check.
- HSN clash.
- SL value.
- LAC boundary.
- DAC value mismatch.
- Syn mismatch.

- Overshoot.
- HW Issues.
- Low Coverage

Solutions for removal of HOSR

Arrange Drive Test:
The best way to find the real issues for HO fail make DT and check layer 3 msg for HO fail. By DT it is very easy to find the fail between cells.

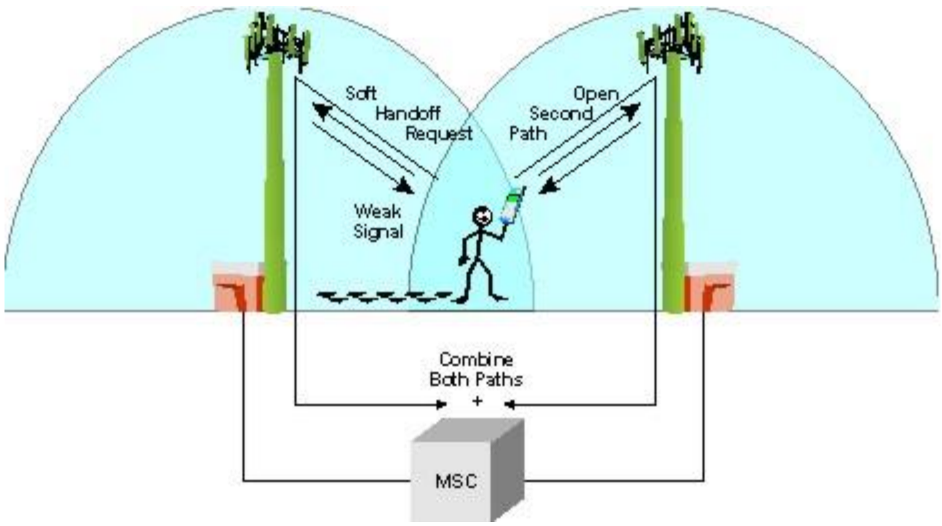
Neighbor Tuning:
Try to retune neighbors
Avoid CO-BCCH-BSIC neighbors.
Avoid extra neighs.
Delete long distance neighs.
Check neighs are defined form both ends.
If there are high fail delete and recreate neighs.

Parameter Check:
1. Retune SL.It can change bw -90,-95,-105.
2. Check HSN.
3. Check SYN.
4. Retune LDR, LUR, IDR, IUR.
5. Retune LMRG, QMRG, PMRG.

DAC value Check:
Check DAC value. If DAC value is high or low tune it at the TH value. It should be 2050.

Optimization Cases

4.1 A Handover Fails Because the BSIC Cannot Be Decoded
4.2 A Handover Fails Because Frequency Sequencing of the MS Is Different from That of the BSC
4.3 A Handover Fails Due to Unreasonable Parameter Configuration
4.4 The Number of Failed Incoming BSC Handovers Increases Because the Handover Request Does Not Contain Class Mark 3
4.5 An Incoming BSC Handover Fails Because the A Interface Phase Flag Is Set Wrongly
4.6 Because the Idle Burst Is Enabled, the Interference Increases, the Receiving Quality Decreases, and the HOSR Becomes Low
4.7 Different HOSRs Resulting from Different Cause Values Contained in the Clear Command Messages Sent by Different Switches.



e. GoS:

Grade of Service (GoS) is defined as the probability that calls will be blocked while attempting to seize circuits. It is written as P.xx blocking factor or blockage, where xx is the percentage of calls that are blocked for a traffic system. For example, traffic facilities requiring P.01 GoS define a 1 percent probability of callers being blocked to the facilities. A GoS of P.00 is rarely requested and will rarely happen because to be 100 percent sure that there is no blocking, you would have to design a network where the caller to circuit ratio is 1:1. Also, most traffic formulas assume that there are an infinite number of callers.

On the loss system, the traffic is carried by network generally lower from real traffic that offered to network by customers. This overload traffic cannot proceed by network and will be loss traffic.

Total this loss traffic is an index from network service quality, is called Grade of Service (GOS), and definition as ratio between loss traffic and offered traffic to network. Offered traffic itself actually is result from total call average from customers and average per call occupation time. Mathematically can explained with equation:

GOS = (A-Y)/A

Because (A-Y) = R = lost traffic

Grade of service = GOS = R/A

Smaller grade of services value, better services that produced. Example, if recommended that grade of services is 0,002, this mean there is two call from every 1000 call or call from every 500 call that offered customer is missing (cannot proceed). If, in one network condition, total customers is same with total server then GOS is same with zero, because every customers that will build a connection will always got success to occupation the server. Network system itself is non-blocking network.

f. Signal receive Level and it's Quality:

A mobile phone signal (also known as reception and service) is the signal strength (measured in dBm) received by a mobile phone from a cellular network (on the downlink). Depending on various factors, such as proximity to a tower, any obstructions such as buildings or trees, etc., this signal strength will vary. Most mobile devices use a set of bars of increasing height to display the approximate strength of this received signal to the mobile phone user. Traditionally five bars are used; see five by five.

Generally, a strong mobile phone signal is more likely in an urban area, though these areas can also have some "dead zones" where no reception can be obtained. Cellular signals are designed to be resistant to multipath reception, which is most likely to be caused by the blocking of a direct signal path by large buildings such as high-rise towers. By contrast, many rural or sparsely inhabited areas lack any signal or have very weak fringe reception; many mobile phone providers are attempting to set up towers in those areas most likely to be occupied by users, such as along major highways. Even some national parks and other popular tourist destinations away from urban areas now have cell phone reception, though location of radio towers within these areas is normally prohibited or strictly regulated, and is often difficult to arrange.

In areas where signal reception would normally be strong, other factors can have an effect on reception, or may cause complete failure (see RF interference). From inside a building with thick walls or of mostly metal construction (or with denserebar in concrete), signal attenuation may prevent a mobile phone from being used. Underground areas, such as tunnels and subway stations, will lack reception unless they are wired for cell signals. There may also be gaps where the service contours of the individual base stations (Cell towers) of the mobile provider (and/or its roaming partners) do not completely overlap.

In addition, the weather may affect the strength of a signal, due to the changes in radio propagation caused by clouds(particularly tall and dense thunderclouds which cause signal reflection), precipitation, and temperature inversions. This phenomenon, which is also common in other VHF radio bands including FM broadcasting, may also cause other anomalies, such as a person in San Diego "roaming" on a Mexican tower from just over the border in Tijuana, or someone in Detroit"roaming" on a Canadian tower located within sight across the Detroit River in Windsor, Ontario. These events may cause the user to be billed for "international" usage despite being in their own country, though mobile phone companies can program their billing systems to re-rate these as domestic usage when it occurs on a foreign cell site that is known to frequently cause such issues for their customers.

The volume of network traffic can also cause calls to be blocked or dropped due to a disaster or other mass call event which overloads the number of available radio channels in an area, or the number of telephone circuits connecting to and from the general public switched telephone network.

With person-to-person calling, the problems can be overcome without loss of anything except time and, perhaps, a few nerves. It is, however, becoming more important that we have the capability to transmit data via these radio waves. Low quality transmissions will result in missed information, confused messages, and, again, dropped calls. In the case of certain financial transactions, this may mean a duplication of the transaction and an error in the charge.

As we begin to depend more and more on this type of technology, it is important that we understand some of the limitations to the cellular technology and how transmission quality can be anticipated and improved. If we understand some of the reasons for signal strength changes, we can better prepare ourselves for the inevitable.

OVERVIEW

Signal quality can vary whether you or the other party are at a desk, walking around, or in a vehicle. Certainly the terrain will have a great affect on the signal if either of you are driving. Buildings, tunnels, hills, etc. will all decrease the signal level. These will affect your signal if you remain in one place but they can be anticipated. You know that there are places in your home or office that are better than others for cell phone reception. But WHY?

1. SURROUNDINGS

We've touched on some of the types of obstructions to good signal quality…hills, buildings. But, building materials, position within the building, clouds, and even trees can all have an impact on your cell phone function. Yes, even the season of the year will change your signal depending upon whether or not the trees and shrubs have leaves. All of this must be considered when trying to improve your transmission.

2. CELLULAR TOWERS

Your distance and direction to the nearest tower will have the greatest effect on your signal. You must be within "range" to use or continue to use your cell phone. This "range" depends upon a couple of factors…the power of the tower transmitter, and the power of the transmitter in you cell phone. The real distance from this tower can vary up to about 5 miles. This, too, would depend upon the height and output of the tower. Keep in mind that it is easier for your phone to hear the tower than it is for the tower to hear your phone. The output of the tower is much greater than that of your phone. The output of your phone varies between 0.2 and 0.6 watts. Depending upon your location (metropolitan, suburban, rural) the tower could have an output of up to 3.0 watts. This is why you will see a signal on your phone (up to 5 bars) but not able to carry on a conversation.

The tower direction is also a factor. The tower "cells" are put together much like a honeycomb in design. This is done to make some sense of the pattern and not have your conversation spread across several towers at a time. This is also a reason that the power output of your phone will vary up to the full 0.6 watts…so you are linked to only one tower at a time. When you are mobile, one tower will "hand you off" to the next tower. This hand off is not supposed to present a problem…but we all know it can.

Another problem presented by the towers is that of 'antenna position'. The antenna that is to cover your location may not be pointed in the correct direction (high, low, or not at all). The antenna may be in the correct position according to the cell requirements. However, if you are in a location that has always had good reception, you must complain to your carrier. It is possible that a technician "bumped" the antenna.

3.**OBSTRUCTION**

Trees, foliage, hills, buildings, and weather all have an impact on the signal quality that you are trying to achieve. Metal will reflect the waves from the tower and can make it impossible for you to make or receive a call from one point in a building, while a few feet away the signal is fine. Your position within a building may also have an impact on the signal strength.

4.TOWER PROBLEMS

As with everything else in this world, the operation of any device is totally dependant upon the weather. Wind, rain, lightening, and ice will all have a detrimental affect on the function of the tower. Once your call is received by the tower, it is sent on its way via computer. Software problems can slow or stop the transmission. Actually, the programming in your phone may well have an intermittent failure.

5. TOWER CONGESTION

Towers that are very busy will also diminish your signal quality, increase your dropped calls, and will even give you a 'busy signal'. Once you are linked to a tower, the transmission should be fine. However, if you move to a tower that is overloaded, the call will be dropped. Rush hour driving usually indicates 'rush hour' for the towers. Sporting events and other large groups will also cause a delay in finding an "open" line.

REMEDIES

External Antenna

There are several antennas available (depending upon whether you are mobile or in a home/office situation). Just getting your signal beyond some of the obstructions can double your range and decrease your dropped calls. An antenna does not increase the output of your signal. It does help to receive and transmit signals that might otherwise be lost due to obstructions.

Signal Booster

This accessory will boost the output of your cell phone. The booster attaches between your cell phone and the external antenna. It can be powered with the cigarette lighter in your vehicle or by using the AC/DC power adaptor for your home/office power. The signal for both transmission and reception will be increased (bi-directional). The booster can be used at home, in your office, or any vehicle (boat, RV, car).

Signal Repeaters

These units are necessary when an external antenna and/or signal booster will not resolve the problem. The office or convention space may be on the interior of the building. The 'repeater' will pick up the signal of the cell phone and 'move' it to the external antenna and directly to the tower. Many convention centers are set up with this feature. Smaller units work well with homes and small office buildings.

POINT OF SALE

Unfortunately, all of the factors pointed out above will play a part in the success of processing a credit card transaction on our apparatus. Dropped calls, busy signals, poor line quality are all problems that must be anticipated and resolved.

In many cases, the standard antenna of our cellular unit will be sufficient. In metropolitan and suburban areas the cell coverage of a major carrier should handle your needs very well. Your location and surroundings will play a large part in this. Certainly, use of the cell unit in a rural situation will require planning and, probably, 'signal enhancement'.

Chapter 3.7

Speech Quality:

Speech quality in mobile communications was carried out by semantic differentiation and external preference mapping, and the developed attributes were mapped to overall quality judgements. A clean speech sample and a speech sample corrupted by car cabin noise from two speakers were processed by different processing chains representing, e.g., transmission of speech over real GSM networks, various standardised speech coders and speech coding with erroneous transmission channels, etc., resulting in a total of 170 samples. The perceptual characteristics of the test samples were described by 18 screened and trained subjects. The final descriptive language with 21 attributes and their rating scales were developed in panel discussions. The scaled attributes were mapped to overall

quality evaluations collected from 30 screened and trained subjects by partial least-squares regression (PLSR)

The quality of speech in mobile communication was studied by making use of semantic differentiation and external preference mapping (Mattila, [1], pp. 169-202, 208-248). Semantic differentiation (Osgood, [2]) was used to extract the perceptual characteristics of speech and background noise, originated from speech processing, which can be used to differentiate between processed speech samples. External preference mapping (Carroll, [3]) was then applied to establish the attributes explaining overall acceptability and their relative importance to quality. Two phonetically rich sentences spoken by a male and a female speaker served as the basic speech samples. These samples were corrupted by a car cabin noise and together with the clean speech samples processed by 41 different processing chains, representative to mobile telecommunication. Additionally, three specific processing chains were only applied to clean speech.

These 170 test samples may be divided into seven main processing categories including, e.g., transmissions through real mobile communication systems, speech coding algorithms, speech enhancement algorithms, tandem connections of speech coders, speech and transmission channel coding, etc. A "listen and describe" technique was used to collect spontaneous verbal descriptions about the test samples from 15 screened and trained subjects (Mattila and Zacharov, [4]). The objective was to identify and verbally describe all the perceptual characteristics of the speech signal and the background noise process. A distinction was made between descriptions originated from processing and the ones that were also present in the original samples so as to separate the perceptual characteristics of the processing chains. In this first collection, a total of about 15100 words were gathered. Since there may be great differences in the ability to verbalize a perception of an audio characteristic, the subjects were informed about the descriptions given by the other subjects in a replicated run of the collection. This was done to ease the verbalization process and broaden the view to the stimuli. A total of about 21100 words were collected in this second collection.

After the two collections, a preliminary grouping of attributes was carried out by identifying words equal in meaning. Round-table discussions were then held to decide whether a description suggested by one subject was equivalent in meaning to that suggested by someone else, and attempts were made to develop the facility to perceive descriptions once their presence was called to subjects' attention. 21 attributes, eleven for speech signal (tense/sharp, bright, mechanic, metallic, nasal/whining, muffled, interrupted, rough, scratching (frequency and intensity), rustle and distant) and nine for background noise (humming, creaking, noisy, low vs. high, bubbling, hissing, boiling, crackling and fluctuating) were finally considered to represent all the perceptually important aspects of the samples. The panel discussions were also used to develop a rating scale with associated anchor words for each attribute to guarantee effective use of the scales.

Further, the subjects were gathered in groups to select an anchor sample for each attribute from a pre-selected set of samples. These samples served as examples for the specific audio characteristics in question. Finally, the intensities of the 21 attributes in the 170 test samples were evaluated by 18 screened and trained subjects in an attribute scaling test. Here, each attribute was scored separately so as to minimize cross-correlation between attributes. Each subject gave 3570 judgements and a total of 64260 judgements were collected. A factorial analysis of variance (ANOVA) was first performed for the whole data to check if the attributes were different. After this, separate ANOVAs were carried out for each attribute to ascertain that the attributes could provide sample differentiation. Before the semantic differentiation, the test samples were evaluated for absolute overall quality by thirty subjects. 1020 judgments were collected from each subject in six repetitions of the test, resulting in a total of 30600 judgments. The acceptability and the attribute data were checked in principal component analyses (PCA) for clusters of subjects. As subjects seemed to share a similar view to the overall quality and attributes, both data sets were averaged over the subjects.

Partial least square (PLS) regression (Martens and Næs, [5], pp. 85-165) was used to map the attribute scaling data to the acceptability data. The attractive property of PLS is that it could take into account both data sets in the regression model, extracting relevant factors from an interpretation and a prediction point of view. Fourteen of the 21 attributes were noticed to be significant by Marten's uncertainty measure (Martens and Martens, [6]) to explain the acceptability. However, eighteen attributes provided the best prediction model but to obtain a simpler model sixteen attributes, achieving roughly the same performance were used to predict acceptability, providing a root mean

square error for prediction of about 6 %. Hereby, it can be stated that attributes describing perceptually important characteristics of processed speech and background noise can be used to predict quality and to give a multidimensional view to quality.

Chapter 4

MAJOR QVOICE STANDARDS & KPI ANALYSIS

4.1 Major QVoice Standards

The proper received level of the signal and ensuring its highest quality is an integral part of quality of service. Received Signal Level (RxLev) is measured in dBm and the received signal power is within a range of -110dBm to -47dBm. Received Signal Quality (RxQual) is measured in percent of received samples allocated on quality levels.

TABLE 1: MAJOR QVOICE STANDARDS [6]

Rx Level (dBm)	Excellent	Good	Fair	Poor	Bad
	-47 to -61	-62 to -71	-72 to -81	-82 to -91	-92 to -110
RxQual	Excellent	Fair	Bad		
	0 to 3	3 to 5	5 to 7		

4.2 KPI Analysis

The following charts will give a broaden information of the results obtained from a test drive operated by Swiss Qual, that mainly emphasizes on the aforesaid performance parameters.

s Rx quality (%)	Grameen Phone	Robi	Banglalink
Excellent(%)	91.73	94.16	91.80
Fair(%)	5.05	2.88	4.54
Bad(%)	3.22	2.96	3.65

To understand the information and comparisons among these three mobile operators in Bangladesh , we will denote Grameen Phone, Robi, and Banglalink as Operator “A” , “B”and Operator C” respectively.

TABLE 3: RX NOISE LEVEL DISTRIBUTION

Operator	Remarks	Value (dBm)
Operator “A” (900 MHz)	Range	-80 to -77.9
	Average	-78.95
Operator “A” (1800 MHz)	Range	-80 to -76.7
	Average	-78.35
Operator “B” (900 MHz)	Range	-79.9 to -75.7
	Average	-77.8
Operator “B” (1800 MHz)	Range	-80 to -76.1
	Average	-78.05
Operator “C” (900 MHz)	Range	-80 to -78.6
	Average	-79.3
Operator “C” (1800 MHz)	Range	-80 to -78.5
	Average	-79.25

TABLE 4 : BENCHMARKING KPIS

KPIs	Uplink			Downlink		
	Operator “A”	Operator “B”	Operator “C”	Operator “A”	Operator “B”	Operator “C”
Gap (%)	6.76	3.67	3.87	2.99	6.39	6.41
Silence (%)	0.06	0.09	0.06	0	0	0
Echo (%)	0	0	0	0	0	0

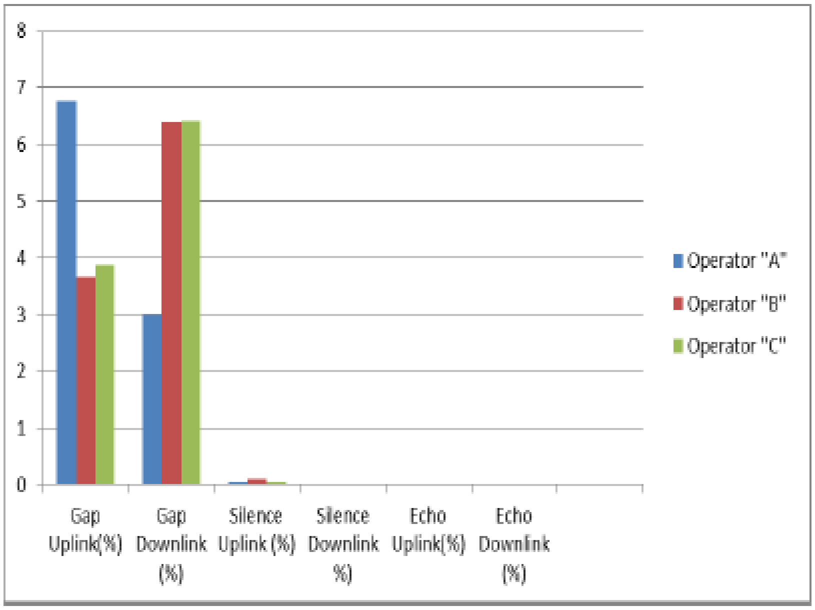


Fig 3: Benchmarking KPIs

TABLE 5: **CALL SETUP SUCCESS RATE**

Operator	Attempts Completed	Attempts Dropped	Total Counts	Success Rate (%)
Operator "A"	82	4	86	95.34
Operator "B"	115	2	117	98.29
Operator "C"	114	5	119	95.80
AVG	103.66	3.66	107.33	96.47

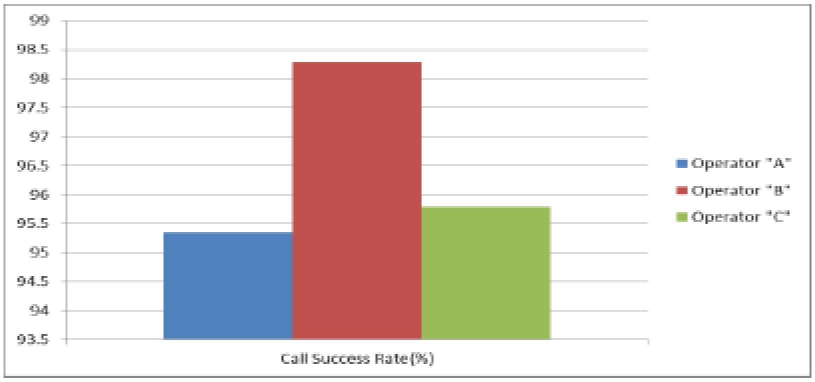


Fig 4: CALL SETUP SUCCESS RATE

TABLE 6 : HANDOVER SUCCESS RATE

Handover	**Data**	**Operator "A"**	**Operator "B"**	**Operator "C"**
Assignment Completed	Avg	212	274	227
	Max	330	330	966
	Min	170	170	140
HO Completed	Avg	313	343	287
	Max	516	626	423
	Min	220	233	186
HO Failed	Avg			599
	Max			640
	Min			566
TCH Assignment Completed	Avg	216	215	195

	Max	266	266	486
	Min	170	170	153
HO not Completed	Avg			
	Max			
	Min			
Assignment Failed	Avg			1280
	Max			1280
	Min			1280
TCH Assignment Failed	Avg			227
	Max			966
	Min			140

TABLE 7: TIMING ADVANCE (T.A)

Average of T.A	Operator A	Operator B	Operator C
Total	0.92	1.07	0.21

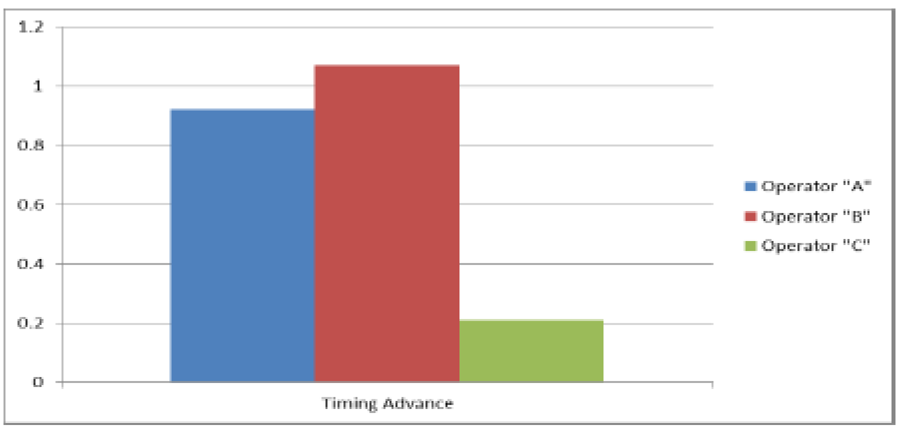


Fig 5: Timing Advance (T.A)

Chapter 5

ANALYSIS

The results show that some of the operators don’t meet the desired level of the performance in different aspects. Thus to ensure that these operators have upgraded their performance, the implementation of the newer technologies is necessary here. These include Transmit Diversity, MIMO etc.

Chapter 6

Solutions

The results show that some of the operators don't meet the desired level of the performance in different aspects. Thus to ensure that these operators have upgraded their performance, the implementation of the newer technologies is necessary here, These include Transmit Diversity, MIMO etc.

Multiple-Input Multiple-Output (MIMO) Optical Wireless Communications

There is growing demand for indoor wireless communication systems with higher bandwidth and higher data rates. However, the crowded radio frequency (RF) spectrum has caused researchers to consider optical wireless systems. In this thesis, optical signals in the visible region of the spectrum are used. White LEDs are used as transmitters as they provide higher signal-to-noise (SNR) levels and a better link budget than the infrared alternative. Typical modulation bandwidths for white LEDs are limited to tens of MHz. Thus, multiple-input multiple-output (MIMO) transmission is considered as a means to increase data rate. The development of the indoor optical wireless MIMO system begins with the geometrical and mathematical analysis of a single-input single-output (SISO) system and a single-input multiple- output (SIMO) system.

The same analysis is then performed for a MIMO system. For the MIMO system, an experimental demonstration using white LEDs and non-imaging receivers are reported. Results include coverage measurements and an SNR analysis. There are limitations using non- imaging receivers, such as coverage limitations and symmetry problems, which cause problem with signal recovery. To improve these limitations, imaging receivers are considered. The design and development of an experimental demonstration of an indoor optical wireless MIMO system with an imaging receiver is presented. The experimental setup consists of a transmitter with a 2 x 2 array of white LEDs and a receiver with a 3 x 3 photo detector array. The system transmits data at a bit rate of 2Mbit/s/channel. Detailed design specifications and optical design are presented. Results show that certain positions within the system coverage area have error-free operation. The BER and SNR analysis shows that the overall BER improves with the overall SNR. In order to exploit the full potential of the system, future work should focus on improving the SNR and BER of the system.

People's ideas about communications have changed completely, nowadays when this subject is mentioned almost everyone thinks of wireless communications. The demand for broadband wireless communications offering greater and greater data rates is endless, and the radio-technology community is trying harder and harder to satisfy this demand. Recently, there was the worldwide launch of 4th generation (4G) systems promising 100 Mbit/s for high mobility communications and up to 1 Gbit/s for stationary or low mobility communications. The key to this technology is the combination of orthogonal frequency-division multiple access (OFDMA) applied to multiple-input multiple-output (MIMO) systems. On the other hand, researchers in wireless optical communications (WOC) are trying to find a way to gain the interest of communication companies by providing new and attractive alternatives to radio communications, as we must not lose sight of the fact that most wireless communications are established inside rooms.

Thus, WOC systems offer some advantages over their radio-frequency (RF) counterparts [15]: they are, theoretically, unregulated and have unlimited1 bandwidth. There is also an inherent security capability, as light (communication) is confined to the room, and there is immunity to multipath fading. However, they are not exempt from drawbacks: strict power limitations due to eye-safety constraints, severe path losses and multipath dispersion and, last but not least, limited maximum achievable signal-to-noise ratio (SNR) due to unavoidable natural and artificial noise sources are the main problems. Over the last few years, OFDM has begun to be proposed for both fiber and wireless optical communications [4] as an effective solution to mitigating inter-symbol interference (ISI) caused by dispersive channels. Furthermore, the frequency-domain channel equalization provided by an OFDM system does not undergo severe complexity penalty when data rates and dispersion increase as opposed to serial time-domain equalizers, and MIMO techniques can be applied to these systems with relative ease. Finally, the complexity of transmitters and receivers is transferred from an analogue to a digital domain by employing Fast Fourier Transform (FFT) and Inverse FFT (IFFT) blocks as demodulators and modulators, respectively. Therefore, all these aspects favour the implementation of OFDM systems in the current digital era. This chapter describes the characteristics of MIMO-OFDM systems applied to WOC, discussing their benefits, but also their drawbacks, as compared with other techniques used in order to obtain high-capacity optical data networks.

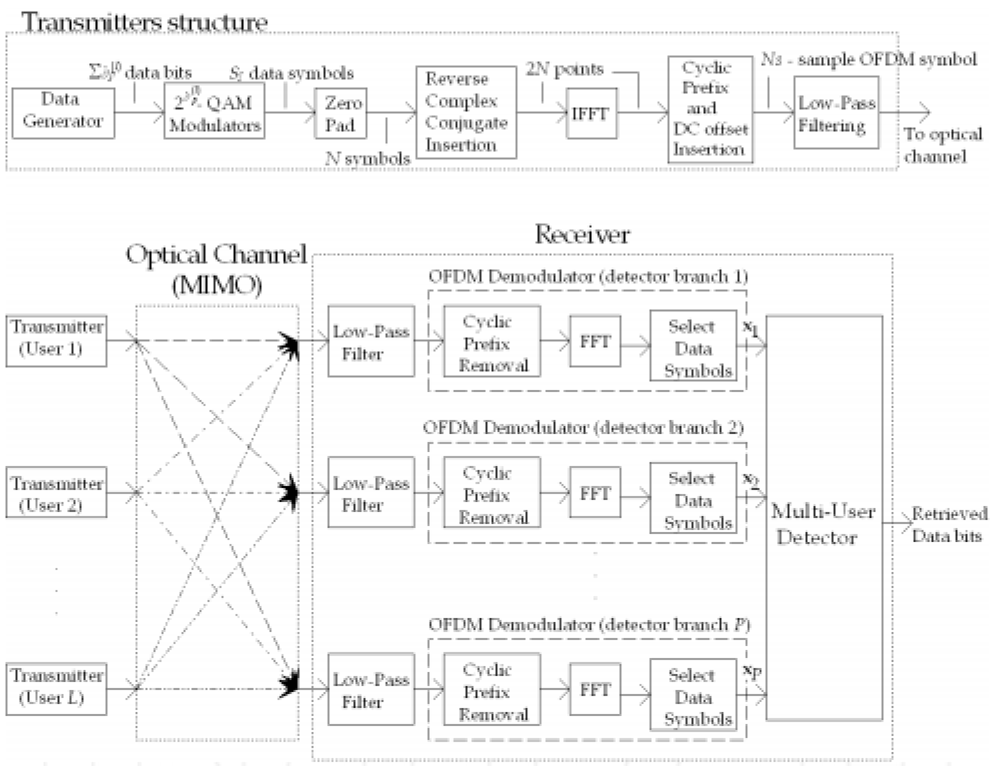
The MIMO optical channel In a typical multi-user application, several emitters can be placed in the room and an angle-diversity receiver, composed of multiple receiving elements oriented in different directions, could be used. By using (7), the contribution of the i th ray emitted by the l th user to the received power reaching each j th branch of the receiver during a certain time interval (p (l) j,i,k, kth time interval) can be computed. The total received power at the j th branch of the optical detector in the kth time interval (width Δt) is computed as the sum of the power of the N(l) j,k rays that contribute in that interval

$$p_{j,k}^{(l)} = \sum_{i=1}^{N_{j,k}^{(l)}} p_{j,i,k}^{(l)}$$

If we consider a normalized receiver responsivity of 1 A/W, the impulse responses h (l) j (t) at all branches (j = 1, . . . , P) due to the l th user (l = 1, . . . , L) are given by:

$$h_j^{(l)}(t) = \sum_{k=0}^{K-1} p_{j,k}^{(l)} \delta(t - k\Delta t)$$

where K = tmax/Δt, and where we have assumed as the time origin the instant when the rays are generated from the emitter. This process must be repeated in order to obtain the different impulse responses between each emitter and each receiving branch in the multi-user scenario



There has been a small amount of research in optical MIMO. Optical channels predominantly use intensity modulation and direct (power) detection, so there is no decor relation in most cases, and it is only in the case of long atmospheric paths where turbulence and scattering is likely to cause this [10]. For shorter range systems [11] shows a MIMO approach to modeling an indoor system and [12] studied the capacity of a MIMO system, and a low speed demonstration is reported.

In work on space-time codes for MIMO is detailed. Reference [14] reports some preliminary experiments with a simple MIMO interconnect. There is also large body of literature on optical interconnects between source and detector arrays.

These examples of parallel free-space optical interconnects typically require precise alignment to map a source to a particular detector or group of detectors and achieve this by design of the physical system. MIMO allows the alignment required for such an interconnect to be achieved 'in the electronics' as it is not necessary that light from a source precisely strikes a single detector.

MIMO techniques can used to 'learn' the channel matrix, thus quantifying the crosstalk between the channels created by each source and detector. This can then be used to estimate the transmitted data. The motivation for using MIMO is therefore not for capacity growth, but to reduce the difficulties in achieving alignment physically by using electronic signal processing. In the application considered here each individual LED has very limited bandwidth, but there are many available for data transmission, but it is not possible to precisely align a detector and receiver array as the receiver moves around the coverage area. MIMO provides an opportunity to do this, and in this paper we consider two 'limiting cases' of receiver types. In the first an array of receivers, each with their own optical concentrator is considered, and in the second an imaging diversity receiver structure is used.

Indoor optical wireless systems often use arrays of transmitters and receivers to achieve high data rates. MIMO offers the potential to use these components to transmit data in parallel between multiple sources and detectors. The slowly varying incoherent communications channel can be measured using straightforward techniques, and resulting channel matrix used to recover data that is transmitted in parallel from a number of sources. In this paper we report results from several preliminary indoor communications experiments. A four channel MIMO system that uses white light LEDs for communication (and associated illumination) is described, as well as experiments in a diffuse environment, using infra-red sources. Design issues, as well as future opportunities and challenges are also discussed.

The use of MIMO techniques in radio channels is a long-standing topic of interest. It has been shown that by increasing the number of antennae in a radio system that a diversity gain can be achieved, i.e., an improvement in the reliability of the system can be had. Alternatively, it has been demonstrated that multiple transmitting and receiving antennae can be used to provide a boost in rate, i.e. a multiplexing gain. In fact, a formal trade-off between diversity and multiplexing gain in MIMO radio channels has been established [1].

The OW MIMO systems discussed in this chapter differ fundamentally from earlier work on radio channels. In particular, only indoor OW channels are considered here and are free of multipath fading. In addition, the signalling constraints imposed by intensity-modulated/direct-detection (IM/DD) systems limits the direct application of theory from radio channels.

Nonetheless, MIMO techniques have been applied to OW channels to yield improvements in reliability and to improve data rates.

The MIMO system utilizes multiple element antennas (MEAs) both on transmit and receive sides of the communication link and thus increases the capacity in a multi-path propagation environment by using Alamouti Codes.

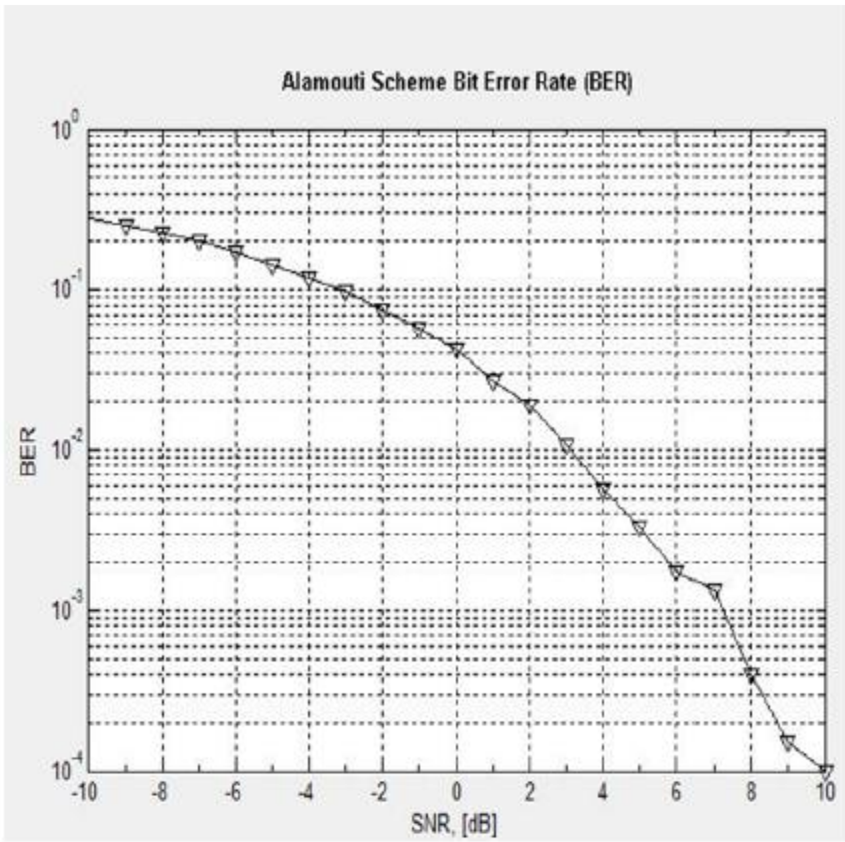


Fig: Alamoti scheme

Transmit Diversity:

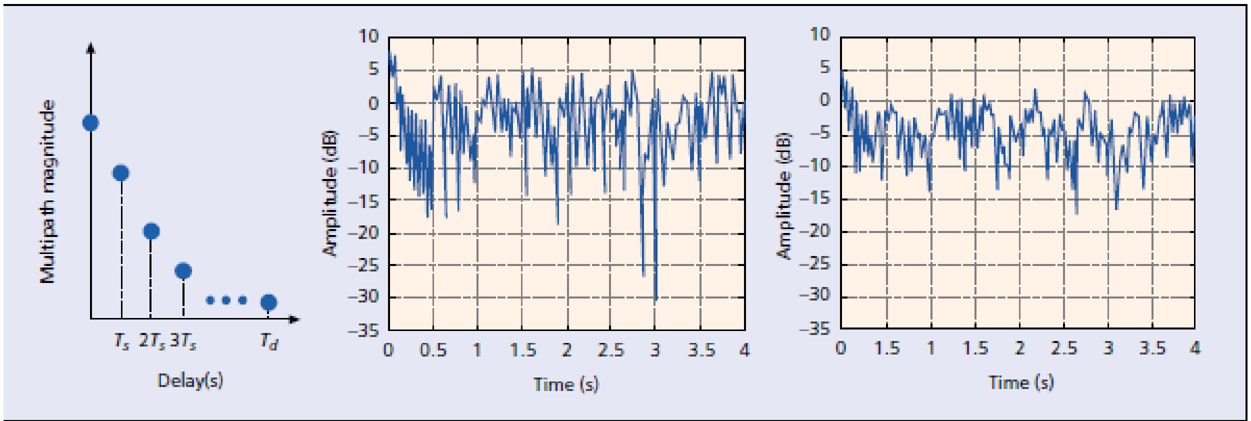


Fig: Transmit Diversity WRT Time

Transmit Diversity diminish the effects of fading by transmitting the same information from two different antennas. The data from the second antenna (Open Loop Antenna 2) is encoded differently to distinguish it from the primary antenna (Open Loop Antenna 1). The user equipment (UE) must be able to recognize that the information is coming from two different locations and properly decode the data.

The increasing demand for higher data rates and higher quality in wireless communications has motivated the use of multiple antenna elements at the transmitter and single antenna at the receiver in a wireless link. Space-time block coding over Rayleigh fading channels using multiple transmit antennas was introduced. In this work Data is encoded using a space-time block code and the encoded data is split into n streams which are simultaneously transmitted using n transmit antennas. The received signal at each receive antenna is a linear superposition of the n transmitted signals perturbed by noise. Maximum likelihood decoding is carried out by dividing the signals transmitted from different antennas. This uses the orthogonal structure of the space-time block code and gives a maximum-likelihood decoding algorithm, which is based only on linear processing at the receiver. Space-time block codes are designed to achieve the maximum diversity order for a given number of transmit and receive antennas subject to the constrain of having a simple decoding algorithm.

This paper presents a simple two-branch transmit diversity scheme. Using two transmit antennas and one receive antenna using QAM modulation technique the performance of OSTBC with

Alamouti is compared with no STBC scheme at lower as well as higher SNRs. This paper evaluates the performance of the system by increasing data lengths in terms of blocks.

Use transmit diversity (tx diversity) to diminish the effects of fading by transmitting the same information from two different antennas. The data from the second antenna (Open Loop Antenna 2) is encoded differently to distinguish it from the primary antenna (Open Loop Antenna 1). The user equipment (UE) must be able to recognize that the information is coming from two different locations and properly decode the data.

The transmit diversity feature uses STTD encoding to differentiate the signals between Open Loop

Antenna 1 (Antenna 1) and Open Loop Antenna 2 (Antenna 2) on the following channels:

- P-CCPCH
- PICH
- DPCH
- HS-PDSCH
- HS-SCCH
- OCNS

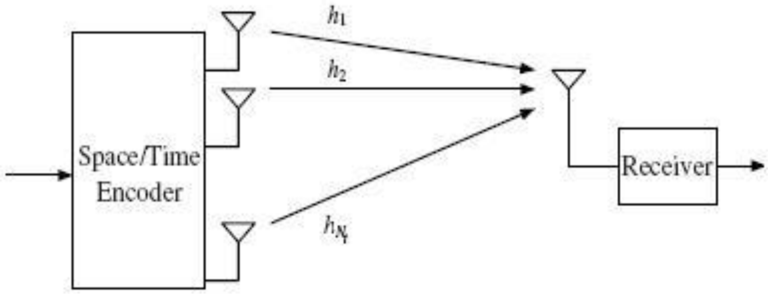
❖ The CPICH is not STTD encoded, but it is affected by transmit diversity. Even though it is transmitted from both antennas, its predefined bit sequence differs between antenna one and antenna two per the 3GPP specifications. The SCH uses TSTD encoding and is phase inverted.

Using both antenna setups at the same time, requires the use of two ESGs and two Signal Studio for 3GPP W-CDMA HSPA software sessions. When using the two antenna setup, you must synchronize the signals between the two ESGs. The following topic discusses the synchronization:

<u>Tx Diversity Signal Synchronization</u>

The following topics describe how to configure the transmit diversity signals:

Step 1: Equipment Setup
Step 2: Setting Up Antenna 2 (ESG 2)
Step 3: Setting Up Antenna 1 (ESG 1)


h_1
h_2
Space/Time
Encoder
h_{N_t}
Receiver

Chapter 7

Appendix

Alamouti Code:

```
function varargout = Alamouti(varargin)
% ALAMOUTI Application M-file for Alamouti.fig
%    FIG = ALAMOUTI launch Alamouti GUI.
%    ALAMOUTI('callback_name', ...) invoke the named callback.
% Last Modified by GUIDE v2.0 25-Jul-2004 12:44:35
if nargin == 0  % LAUNCH GUI
fig = openfig(mfilename,'reuse');
% Use system color scheme for figure:
set(fig,'Color',get(0,'defaultUicontrolBackgroundColor'));
% Generate a structure of handles to pass to callbacks, and store it.
handles = guihandles(fig);
guidata(fig, handles);
if nargout > 0
varargout{1} = fig;
end
elseif ischar(varargin{1}) % INVOKE NAMED SUBFUNCTION OR CALLBACK
try
if (nargout)
[varargout{1:nargout}] = feval(varargin{:}); % FEVAL switchyard
else
feval(varargin{:}); % FEVAL switchyard
end
catch
```

```
disp(lasterr);

end

end

% -----------------------------------------------------------------

function varargout = Start_Callback(h, eventdata, handles, varargin)

%Get sytem parameters from GUI

N=str2double(get(handles.N_input,'String')); %total number of symbol pairs to be transmitted (should be at least 10 times more than expected 1/min(BER))

M=str2double(get(handles.M_input,'String')); %PSK order (must be a power of 2): 2, 4, 8 etc'

Tx=str2double(get(handles.Tx_input,'String')); %number of Tx elements, must be 2

Rx=str2double(get(handles.Rx_input,'String')); %number of Rx elements

SNR=eval(get(handles.SNR_input,'String')); %SNR in dB, average received power at one Rx element over the average noise power at that element

%Simulation parameters for the verson without GUI

%N=1000; %total number of symbol pairs to be transmitted (should be at least 10 times more than expected 1/min(BER))

%M=2;   %PSK order (must be a power of 2): 2, 4, 8 etc'

%SNR=0:10; %SNR in dB, average received power at one Rx element over the average noise power at that element

%Tx=2; %number of Tx elements, must be 2

%Rx=1; %number of Rx elements

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

%Resets the generators to their initial state

randn('state',0); %Remove it if you want a random start of the randn generator

rand('state', 0); %Remove it if you want a random start of the rand generator

%Monte-Carlo

for k=1:length(SNR)
```

```
%Toss pairs of uniformly distributed MPSK symbols with power 1/2
A=floor(M*rand(2,N)); %transmitted alphabet
st=exp(j*2*pi/M*A)/sqrt(2); %transmitted symbols
%Simulate equivalent matrix of impulse noise
%Noise power caculation
snr=10^(SNR(k)/10); %just translate SNR from dB to times
sig1=0.5/snr; %the sigma square of the noise
Ns=sqrt(sig1)*(randn(2*Rx,N)+j*randn(2*Rx,N)); %noise matrix
%Transceiver
for n=1:N
%Toss the channel complex coefficients
H=[]; %equivalent channel matrix initialization
for r=1:Rx
h=(randn(1,2)+j*randn(1,2))/sqrt(2); %Rayleigh channel
%h=ones(1,2); %flat channel
%Equivalent channel matrix:
%h(1) - from first Tx to current Rx; h(2) - from second Tx to current Rx
H=[H; h(1) h(2); h(2)' -h(1)'];
end %m
sr(:,n)=H'*H*st(:,n)+H'*Ns(:,n); %received symbols
end %n
%ML detection
ang=angle(sr); %received angles
B=mod(round(ang/(2*pi/M)),M); %received alphabet
%BER estimation (for the Gray code constellations)
BER(:,k)=sum(sum(xor(A-B,0)))/2/N/log2(M);
```

BPSK Modulation with alamouti Code

```
clear

N = 10^6; % number of bits or symbols

Eb_N0_dB = [0:25]; % multiple Eb/N0 values

for ii = 1:length(Eb_N0_dB)

   % Transmitter

   ip = rand(1,N)>0.5; % generating 0,1 with equal probability

   s = 2*ip-1; % BPSK modulation 0 -> -1; 1 -> 0

   % Alamouti STBC

   sCode = zeros(2,N);

   sCode(:,1:2:end) = (1/sqrt(2))*reshape(s,2,N/2); % [x1 x2  ...]

   sCode(:,2:2:end) = (1/sqrt(2))*(kron(ones(1,N/2),[-1;1]).*flipud(reshape(conj(s),2,N/2))); % [-
x2* x1* ....]

   h = 1/sqrt(2)*[randn(1,N) + j*randn(1,N)]; % Rayleigh channel

   hMod = kron(reshape(h,2,N/2),ones(1,2)); % repeating the same channel for two symbols

   n = 1/sqrt(2)*[randn(1,N) + j*randn(1,N)]; % white gaussian noise, 0dB variance

   % Channel and noise Noise addition

   y = sum(hMod.*sCode,1) + 10^(-Eb_N0_dB(ii)/20)*n;

   % Receiver

   yMod = kron(reshape(y,2,N/2),ones(1,2)); % [y1 y1 ... ; y2 y2 ...]

   yMod(2,:) = conj(yMod(2,:)); % [y1 y1 ... ; y2* y2*...]

   % forming the equalization matrix

   hEq = zeros(2,N);

   hEq(:,[1:2:end]) = reshape(h,2,N/2); % [h1 0 ... ; h2 0...]

   hEq(:,[2:2:end]) = kron(ones(1,N/2),[1;-1]).*flipud(reshape(h,2,N/2)); % [h1 h2 ... ; h2 -h1 ...]

   hEq(1,:) = conj(hEq(1,:)); %  [h1* h2* ... ; h2 -h1 .... ]

   hEqPower = sum(hEq.*conj(hEq),1);

   yHat = sum(hEq.*yMod,1)./hEqPower; % [h1*y1 + h2y2*, h2*y1 -h1y2*, ... ]
```

```
    yHat(2:2:end) = conj(yHat(2:2:end));
    % receiver - hard decision decoding
    ipHat = real(yHat)>0;
    % counting the errors
    nErr(ii) = size(find([ip- ipHat]),2);
end
simBer = nErr/N; % simulated ber
EbN0Lin = 10.^(Eb_N0_dB/10);
theoryBer_nRx1 = 0.5.*(1-1*(1+1./EbN0Lin).^(-0.5));
p = 1/2 - 1/2*(1+1./EbN0Lin).^(-1/2);
theoryBerMRC_nRx2 = p.^2.*(1+2*(1-p));
pAlamouti = 1/2 - 1/2*(1+2./EbN0Lin).^(-1/2);
theoryBerAlamouti_nTx2_nRx1 = pAlamouti.^2.*(1+2*(1-pAlamouti));
close all
figure
semilogy(Eb_N0_dB,theoryBer_nRx1,'bp-','LineWidth',2);
hold on
semilogy(Eb_N0_dB,theoryBerMRC_nRx2,'kd-','LineWidth',2);
semilogy(Eb_N0_dB,theoryBerAlamouti_nTx2_nRx1,'c+-','LineWidth',2);
semilogy(Eb_N0_dB,simBer,'mo-','LineWidth',2);
axis([0 25 10^-5 0.5])
grid on
legend('theory (nTx=1,nRx=1)', 'theory (nTx=1,nRx=2, MRC)', 'theory (nTx=2, nRx=1,
Alamouti)', 'sim (nTx=2, nRx=1, Alamouti)');
xlabel('Eb/No, dB');
ylabel('Bit Error Rate');
title('BER for BPSK modulation with Alamouti STBC (Rayleigh channel)');
```

Chapter 8

Conclusion

In this paper we have used Swiss Qual dive test data to create a analysis of present GSM QoS condition of Bangladesh.As the numbers of subscribers are increasing here day by day, so the Mobile Network Operators have to improve KPIs to cope up with the extended subscriber’s pressure. That’s whywe have analyzed all the major QVoice Parameters. And we have observed the necessity of MIMO and Transmit Diversity to keep the Network Condition sound and smooth Besides fading effect is a major part in MIMO which is on process for our future analysis.

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