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Operating System Based Perceptual Evaluation of Call Quality in Radio Telecommunications Networks

Development of call quality assessment at mobile terminals using the Symbian operating system, comparison with traditional approaches and proposals for a tariff regime relating call charging to perceived speech quality

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Dedicated to my parents, brothers, sisters, wife and children.

Acknowledgements

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Keywords

Speech quality measurement; Call quality; Signal strength; Mobile cellular telecommunication networks; Non-dedicated and heterogeneous network; QMeter®; SM (signal meter); BM (bandwidth meter)

Abstract

Call quality has been crucial from the inception of telecommunication networks. Operators need to monitor call quality from the end-user's perspective, in order to retain subscribers and reduce subscriber "churn". Operators worry not only about call quality and interconnect revenue loss, but also about network connectivity issues in areas where mobile network gateways are prevalent. Bandwidth quality as experienced by the end-user is equally important in helping operators to reduce churn.

The parameters that network operators use to improve call quality are mainly from the end-user's perspective. These parameters are usually ASR (answer seizure ratio), PDD (post-dial delay), NER (network efficiency ratio), the number of calls for which these parameters have been analyzed and successful calls. Operators use these parameters to evaluate and optimize the network to meet their quality requirements.

Analysis of speech quality is a major arena for research. Traditionally, users' perception of speech quality has been measured offline using subjective listening tests. Such tests are, however, slow, tedious and costly. An alternative method is therefore needed; one that can be automatically computed on the subscriber's handset, be available to the operator as well as to subscribers and, at the same time, provide results that are comparable with conventional subjective scores. QMeter® – a set of tools for signal and bandwidth measurement that have been developed bearing in mind all the parameters that influence call and bandwidth quality

experienced by the end-user – addresses these issues and, additionally, facilitates dynamic tariff propositions which enhance the credibility of the operator.

This research focuses on call quality parameters from the end-user's perspective. The call parameters used in the research are signal strength, successful call rate, normal drop call rate, and hand-over drop rate. Signal strength is measured for every five milliseconds of an active call and average signal strength is calculated for each successful call. The successful call rate, normal drop rate and hand-over drop rate are used to achieve a measurement of the overall call quality. Call quality with respect to bundles of 10 calls is proposed.

An attempt is made to visualize these parameters for better understanding of where the quality is bad, good and excellent. This will help operators, as well as user groups, to measure quality and coverage.

Operators boast about their bandwidth but in reality, to know the locations where speed has to be improved, they need a tool that can effectively measure speed from the end-user's perspective. BM (bandwidth meter), a tool developed as a part of this research, measures the average speed of data sessions and stores the information for analysis at different locations.

To address issues of quality in the subscriber segment, this research proposes the varying of tariffs based on call and bandwidth quality. Call charging based on call quality as perceived by the end-user is proposed, both to satisfy subscribers and help operators to improve customer satisfaction and increase average revenue per user. Tariff redemption procedures are put forward for bundles of 10 calls and 10 data sessions. In addition to the varying of tariffs, quality escalation processes are proposed. Deploying such tools on selected or random samples of users will result in substantial improvement in user loyalty which, in turn, will bring operational and economic advantages.

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Abbreviations used in the text

1G	First generation
2G	Second generation
2.5G	2.5 generation
3G	Third generation
3GPP	Third generation partnership proposal
4G	Fourth generation
A/D	Analogue to digital
AAA	Authentication, authorization and accounting
ACELP	Algebraic code excited linear predictive coder
AMC	Adaptive modulation and coding
AMPS	Advanced mobile phone system
AMR	Adaptive multi-rate
AP	Access point
ARIB	Association of Radio Industry and Business
ARQ	Automatic repeat request
ASK	Amplitude shift keying
ASN	Access service network
ASP	Application service provider
ASR	Answer seizure ratio
ATM	Asynchronous transfer mode
AUC	Authentication centre
AWGN	Additive white Gaussian noise

BCH	Bose, Chaudhuri, Hocquenghen
BEP	Bit error probability
BER	Bit error rate
BM	Bandwidth meter
BS	Base station
BSD	Bark spectral distortion
BSS	Business support systems
BTR	Bit throughput rate
C/I	Channel interference ratio
C/N	Carrier to noise ratio
CDMA	Code division multiple access
CDPD	Cellular digital packet data
CEPT	European Conference of Postal and Telecommunications Administrations
CLDC	Connected limited device configuration
CN	Core network
CPE	Customer premises equipment
CTIMIT	Cellular version of TIMIT
DBMS	Database management system
DCS	Digital cellular system
DECT	Digital European cordless telecommunications
DHCP	Dynamic host control protocol
DSL	Digital subscriber line
DTX	Discontinuous transmission
EDGE	Enhanced data rates for GSM evolution
EIR	Equipment identity register

ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission
FCC	Forward control channel
FDD	Frequency division duplex
FDMA	Frequency division multiple access
FM	Frequency modulation
FTTB	Fibre-to-the-building
FTTH	Fibre-to-the-home
GGSN	Gateway GPRS support node
GPRS	General packet radio service
GSM	Global system for mobile communications
HLR	Home location register
HMM	Hidden Markov models
HSCSD	Speed circuit switched data
HTTP	Hypertext transfer protocol
IEEE	Institute of Electrical and Electronics Engineers
IM	Inter-modulation
IP	Internet protocol
ISI	Inter-symbol interference
ITU	International Telecommunication Union
ITU-T	ITU - telecommunication standardization sector
LOS	Line-of-sight
LPC	Linear predicative coding
LPS	

LTE	Long term evolution
LTP	Long term predictor
MBAN	Medical body area network
MBSD	Modified BSD
MIDP	Mobile information device profile[
MIME	*Multipurpose internet mail extensions*
MIP-HA	Mobile IP home agent
MNB	Measuring normalizing blocks
MPEG	Moving pictures experts group
MOS	Mean opinion score
MS	Mobile station
MSC	Mobile switching centre
MSISDN	Mobile station integrated services digital network
MT	Mobile terminal
MUX	Multiplexer
NBAP	Node-B application protocol
NER	Network efficiency ratio
NLOS	Non line-of-sight
NSP	Network service provider
OFDM	Orthogonal frequency-division multiplexing
OFDMA	Orthogonal frequency division multiple access
OS	Operating system
OSI	Open systems interconnection
OSS	Operations support systems

PAMS	Perceptual analysis measurement system
PANs	Personal area networks
PC	Personal computer
PCS	Personal communications services
PDA	Personal data assistant
PDD	Post-dial delay
PESQ	Perceptual evaluation of speech quality
PLMN	Public land mobile network
PSQM	Perceptual speech quality measure
PSTN	Public switched telephone network
QoS	Quality of service
RCC	Reverse control channel
RF	Radio frequency
RNC	Radio network controller
RN LFCC	Root-normalized linear frequency cepstral coefficients
ROM	Read only memory
RPE	Regular pulse excitation
RPP	Regular pulse pattern
RSSI	Received signal strength indicator
SAT	Supervisory audio tone
SGSN	Serving GPRS support node
SIM	Subscriber identity module
SIP	Session initiation protocol
SM	Signal to noise and interference ratio
SMS	Short message service
SNR	Signal to noise ratio

SNIR	Signal meter
SONET	Synchronous optical network
SQI	Speech quality index
TACS	Total access communication system
TDD	Time division dúplex
TDM	Time division multiplexing
TDMA	Time division multiple Access
TD-SCDMA	Time division-synchronous code-division multiple access
TE	Terminal equipment
TIA	Telecommunications Industry Association
TIMIT	A phonetic database that was recorded at Texas Instrument (TI) and transcribed at Massachusetts Institute of Technology (MIT)
TOSQA	Telecommunication objective speech quality assessment
TTA	Telecommunications Technology Association
UE	User equipment
UMTS	Universal mobile telecommunications system
UTRAN	UMTS terrestrial radio access network
UWB	Ultra-wideband
VHE	Virtual home environment
VLR	Visitor location register
VoIP	Voice over internet protocol
WCDMA	Wideband CDMA
WiFi	Wireless fidelity

WiMax	Worldwide inter-operability for microwave access
WLAN	Wireless local area network
WPAN	Wireless personal area network

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List of related publications by the author of this thesis

- A. Aburas, J. G. Gardiner and Zeyad Al-Hokail, “Symbian Based Perceptual Evaluation of Speech Quality for Telecommunication Networks”, Sixth International Conference on Computing, Communications and Control Technologies (CCCT) 2008, Orlando, Florida, USA.
- A. Aburas, J. G. Gardiner and Zeyad Al-Hokail, “Perceptual Evaluation of Speech Quality Implementation Using a Non-Traditional Symbian Operating System”, Fifth IEEE-GCC (Institute of Electrical and Electronics Engineers-General Communication Channel) Conference on Communication and Signal Processing, 17-19 March 2009, Kuwait City, Kuwait.
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CHAPTER 1

Introduction and overview

1.1 The problem

Until the introduction of cellular radio systems in the 1970s, mobile communication was confined to closed user groups such as the emergency services, taxi companies and public utilities. They used simple systems to keep in touch with personnel. Such systems had limited interface with the fixed wire-line network, with dedicated fixed links between control facilities and the transmission and reception base stations on hill-top sites. A limited connection to the fixed network was available but the user equipment was bulky and expensive so that the user community was confined to a small number of business and public sector subscribers.

The situation now is completely different with the majority of the population in the developed world being users of mobile radio and expecting to have access to as many as possible of the services available in the fixed network. Additionally, a range of new technologies has been introduced to cater for the demands of huge numbers of users and the continually growing requirement for transmissions needing to deliver vast amounts of data for downloading entertainment and so on.

Where the market for mobile radio user equipment is largely saturated, network operators must compete to win and retain customers. Given freedom of choice, subscribers will access different networks to find the most appropriate for their needs. The process of changing service provider is usually referred to as “churn” and network operators and service providers will try to minimize this in order to maintain their revenues.

There are many ways in which users experience the effectiveness of the network they are connected to. Moreover, different services are more or less sensitive to the continuity of connection. Given that many services are now delivered by packet switching, rather than circuit switching, high bit error rates can often be disguised by various strategies which are discussed in Chapter 2. But the one sensitive performance criterion, as far as users are concerned, is voice quality [1].

There are two aspects to this. One is the subjective experience of holding a voice conversation when users expect the same quality of connection as calls on the fixed network. The second is the problem of loss of connection during a call. Failure to make a connection in the first instance is annoying, but loss of connection during a call is generally regarded as much more so. It is therefore appropriate to consider how the end-to-end quality of a radio connection is influenced by the various processes which a call must go through.

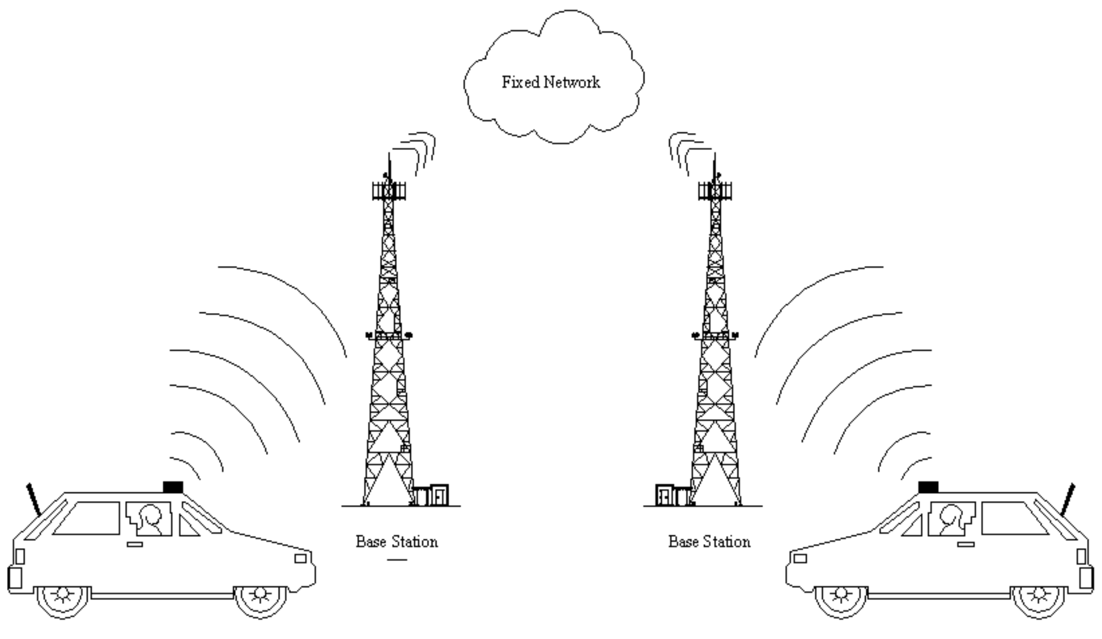


Figure 1: Mobile conversation through fixed network

In Figure 1 a typical user is initiating a call. If he or she is in a vehicle, there might be a short-range connection from a simple head-set to the fixed transponder in the vehicle. This latter unit sets up a call to the network operator's system and if a successful connection is established through the operator's base station the call proceeds into the fixed infrastructure. The network operator's system interfaces with the national and international fixed networks which will usually involve long distance transmission on optical fibre. At the recipient end, a similar series of processes takes place and connection is established.

So where are the weak links in this chain of processes? Once a call is established in the fixed elements of the system, bit error rates (BER) are very small – typically 10^{-7} or better. In the vehicle at the sending end the connection to the vehicle transponder will be very good as the physical relationship between head-set and transponder does not change during the call regardless of the movement of the vehicle. This leaves the connection from vehicle to operator base station. Almost invariably this is the weakest link.

1.2 Continuity of connection

There is an extensive literature dealing with the characteristics and behaviour of the mobile radio channel. This area of work is covered in many publications. A full treatment is available in [2].

There are two principal aspects to note in the context of this thesis. Thanks to shadowing of the signal and multi-path effects, the received signal to a moving mobile and from a moving mobile fluctuates over a wide range [3]. This means that occasionally the received signal at a mobile may fall below a threshold for satisfactory reception and the connection is lost.

The second relates to co-channel interference. It is a fundamental aspect of cellular system operation that the spectrum in any given cell is reused many times in surrounding cells. It follows that a mobile in any cell is subject to interference from the reuse of its allocated spectrum in surrounding cells. Given the variability in signal strength referred to above, there is clearly the possibility that the total received signal at a mobile may be well above the threshold required for reception but this may be mainly interference rather than wanted signal, and again the connection is lost [3].

Advances in technology have made it possible for network operators to support both a huge number of users for basic voice communication services and also the demands of users for access to vast amounts of data, particularly in the entertainment service sector [3]. These advances have taken place in many different areas. To combat the variability of received signal, diversity techniques have been available for many years [4]. More recently, as the processing power in mobile terminals has increased, more powerful combinations of spatial, time and coding diversity techniques have been implementable resulting in “space-time” coding [5].

The outcomes of these advances have also made possible new techniques in the physical layer in which multiple input and output (MIMO) antenna configurations have made further contributions to support growing user communities and growing demands for services [6].

In addition to these aspects, standards-making activities internationally have approved many ways to take advantage of technological advances. These are addressed in more detail in Chapter 2, where the significance of moving from circuit switching to packet switching is set out. However, the question that remains as far as network operators are concerned is of how to maximize the effectiveness of their networks taking advantage of current technology and anticipated developments.

1.3 System planning

Many researchers have contributed to evolving strategies for optimizing the distribution and location of base stations to give uninterrupted coverage to and from mobile terminals. Published work is available in many sources, typically [7], in which the basis of planning is statistical.

Recalling the comments above about how annoying subscribers find dropped calls, once established, the question arises as to how to design a network to best effect; that is, where to position base stations, what antenna configuration to deploy and so on. A further factor is the relationship between outdoor and indoor system implementation. In this consideration, typical research output appears in [8].

Returning to the issue of the separate propagation and interference characteristics of the mobile environment, in [9] it is argued that signal strength variations and outages due to co-channel interference are essentially uncorrelated. Therefore, in assessing the performance of a practical network deployment, both aspects must be measured together. Signal coverage achievable in relation to various terrain features, density of buildings in urban environments, propagation over water and so on are well covered in [2]. Outages have also been considered in [10]. From a practical point of view, field strength predictions can be encapsulated in software and a number of successful planning tools have been available commercially for a number of years, for example [10].

However, there remains a difficulty in relation to predictions expressed in probabilities, since while such predictions give overviews of system performance they do not give specific data in relation to the positions of a mobile on the ground. As the search for more resources for mobile services presses for higher frequency windows in the spectrum, so the range of environmental parameters that are significant also expands. To give accurate predictions of field strength at a mobile terminal, calculations would have to take account of such factors as:

- operating frequency;
- terrain variations – contours, water features;
- building details – height, separation;
- building materials – roughness/smoothness of walls, area of glass; and
- distribution of vegetation – height and spacing of trees.

Bearing in mind that many of these features would be subject to frequent and unrelated changes, it is evident that field strength prediction at the level of detail needed by network operators is not a realistic option. What network operators require is knowledge of how mobiles experience the environment they operate in – not only received wanted signals but also the level of interference and the occurrence of failed hand-overs which result in call outages.

Fortunately, means to achieve this are available since many manufacturers of mobile equipment have adopted the Symbian operating system as standard and this can be used, as will be seen in later chapters of this thesis, to provide the detailed information that network operators need to optimize their systems. While Android has overtaken Symbian now, it is evident that a similar strategy could be applied to both operating systems.

1.4 Organization of the thesis

The objective of the thesis is to address and evaluate the call quality and bandwidth quality of the mobile operator's network from the end-user's perspective. The research demonstrates that an automatic call quality assessment can be performed on every telephone call received by a mobile end-user, giving a call quality score for each call along with call quality statistics for a bundle of calls based on various call quality parameters. This gives subscribers a practical way to compare the performance of different operators. Additionally, operators may wish to give their employees a version of this software to facilitate internal network auditing.

Subsequent to this chapter, the thesis deals with topics as follows:

- Chapter 2 considers the evolution of mobile communications technology.
- Chapter 3 examines strategies for supporting quality of service.
- Chapter 4 looks at voice quality.
- Chapter 5 gives an overview of the Symbian operating system's architecture and the proposed QMeter®.
- Chapter 6 presents generic QMeter® algorithms, with breakdown flow charts of different modules.
- Chapter 7 reports experimental results from implementation of the proposed system.
- Chapter 8 describes the application of QMeter® research at user, operator and regulator levels.
- Chapter 9 proposes a strategy to enable network operators to charge subscribers a variable rate per call, depending on the quality of service delivered.
- Chapter 10 draws conclusions and makes recommendations for further work.

CHAPTER 2

Mobile communications technology evolution

2.1 Historical perspective

In order to appreciate the challenges currently faced by cellular network operators and service providers it is important to review the way in which cellular systems have evolved over the last 40 years. Although this time scale is short in comparison with what has been required for the development of modern motor cars or aircraft, it has nevertheless been sufficient for the complete transformation of mobile telecommunication technology.

Historically, mobile communication systems have been developed in stages, often referred to as "generations", in which progress from one generation to the next has coincided with major advances in technology and service capability. The targets in progressing through the different generations have been fourfold:

1. to use the limited radio spectrum available to cellular radio to provide services to a continually expanding population of users;
2. to progress from a predominantly voice orientated service to higher data rate provision;
3. to take advantage of rapid advances in integrated circuit technology to deploy increasingly sophisticated signal processing techniques to meet objectives (1) and (2);

4. to make major system parameters adaptable in response to the varying radio propagation conditions in the transmission path and the varying demands for radio resources to support differing, often high bit rate, services to cater for differing user needs.

Returning to the generation concept, the first true cellular radio system was developed in the USA and subsequently modified for implementation in the UK. The system used analogue technology with frequency modulation for both voice traffic and control signalling. The significant difference between the USA system, the “advanced mobile phone system” (AMPS) and the UK system, the “total access communication system” (TACS) was the channel bandwidth used – 30kHz channels in the USA, 25kHz in the UK.

The first generation system immediately attracted an enthusiastic response from potential users and it was apparent, even before the TACS system had been rolled out across the UK, that analogue technology could not meet the user capacity demand which, by then, had been clearly identified. A further factor had a great influence over the next step. Other basic cellular systems had been developed to varying degrees elsewhere in Europe and Scandinavia, but they were all mutually incompatible so that a cellular handset designed to one system could be used only in its country of origin. Given the extent to which European unification, both political and economic, had been achieved, it was evident that whatever new system was to be introduced to meet the expectations of users, it would have to be a single system common to all European countries. Further, the limitations of analogue technology were evident. Accordingly a co-ordinated European research and development programme was established with a secretariat based in Paris to oversee the development of a common European digital system. The secretariat was named the *Groupe Special Mobile* (GSM). Subsequently the system specification that evolved has been adopted in many countries and GSM has come to mean “global system for mobile” communications.

Naturally, the constraints imposed by analogue technology had to be overcome by using the signal processing power of digital systems. Several major companies – including Nokia and Ericsson – developed competing systems for adoption as the GSM standard. The systems are compared in Table 1.

System and originator	**Multiple access**	**Carrier separation**	**Channels per carrier**
CD 900, SEL and others	WB-TDMA	6MHz	60
MATS-D, Philips	TDMA/CDMA	2MHz	60
S900D, Bosch	NB-TDMA	250kHz	6
SFH900, Matra	FH	150kHz	3
DMS 90, Ericsson	NB-TDMA	300kHz	10
DPM, ELab	NB-TDMA	600kHz	24
MAX, STA	NB-TDMA	50kHz	4

Table 1: Competing systems for adoption as the GSM standard

Most are recognizable as versions of narrowband time-division multiple access (TDMA), the exception being the proposal from Matra, which used frequency hopping, and the Philips MATS-D system. The latter proposed the use of code-division multiple access (CDMA), which ultimately found application in the more recent third generation (3G) systems. At the time, this technology was considered to be too advanced for implementation in GSM.

As was to be expected, given the expertise in the competing companies, all the systems performed to expectations, but it was decided that none of the individual systems was optimum for a single Europe-wide system and a specification was drawn up taking the best elements of the trialed systems. This became the GSM standard and represented the second generation in cellular mobile system evolution.

Once the major step from analogue to digital system technology had been accomplished, many innovations became possible. The next major step was the implementation of CDMA in the radio transmission path. CDMA technology had already been available in military communications applications when the GSM system was in development. But it was judged to be too advanced and expensive for deployment in GSM at that time. Thus the basic decision was taken to use TDMA as the GSM standard. However, the advances in integrated circuit technology referred to above had transformed the way of transmitting digital frequencies simultaneously over the same carrier frequency by dividing the signal into different time slots, thus allowing several users to share the same frequency channel. CDMA now became feasible as the standard transmission technology in the third generation (3G) system. Additionally, because of the inherent properties of CDMA, it became possible to implement the adaptability in system parameters which formed the fourth target referred to above. In the development phase this was universally accepted as the third generation system and became known as the universal mobile telecommunications system (UMTS).

2.2 Data services

While the progress from one generation to another had tackled the user capacity requirement with great success, the problem of providing users with high-rate data services needed to be addressed by deploying a further layer in the system infrastructure. In particular, means had to be found to incorporate support for the universally accepted data standard in fixed networks – i.e. packet switching.

Initially, attempts were made to introduce packet switched data into the US AMPS system by deploying cellular digital packet data (CDPD) in the 30kHz channels, but only modest data rates could be supported (19.2kbps). In the GSM system a general packet radio service (GPRS) was deployed as a packet overlay network but used the standard GSM frame structure. For voice traffic in GSM, each speech channel uses one of eight time slots in the transmitted frame and the frame transmission rate is set to occupy the allocated bandwidth of 200kHz per carrier. Clearly, if a GSM carrier were committed to data traffic, all eight time slots could be used for packet data, and if the modulation constraints of voice connections could be relaxed, higher order modulation schemes could be used.

These possibilities were realized in the development of a GSM enhancement known as “enhanced data rates for GSM evolution” (EDGE). With these advances, substantially higher data rates could be possible and 547.2kbps was achievable under favourable conditions.

While this represented a significant advance, it was clear that much needed to be done to meet increasingly demanding requirements.

2.3 The TACS system

2.3.1 Functionality

Although, in comparison with subsequent system technologies, the total access communication system (TACS) is relatively unsophisticated, nevertheless it embodies the essential elements of all subsequent systems. Mobiles need to be able to initiate communication wherever they are; calls need to be routed to mobiles wherever they are; mobiles must distinguish between wanted messages and co-channel interference – a fundamental issue since frequency reuse is the basis of the cellular approach; a mobile engaged in a call must be able to continue connection as it moves from cell to cell, i.e. hand-over.

In each cell, a base station must provide the radio resources to support mobiles "camped" in its coverage area. Generally the radio standards used for this are incompatible with the standards used in the fixed network so the base stations need to from a separate network, a public land mobile network (PLMN), to manage the mobiles and provide an interface with fixed public telecommunications resources. The PLMN must have mobile switching centres (MSCs) to handle the generated traffic resulting in the configuration of Figure 2.

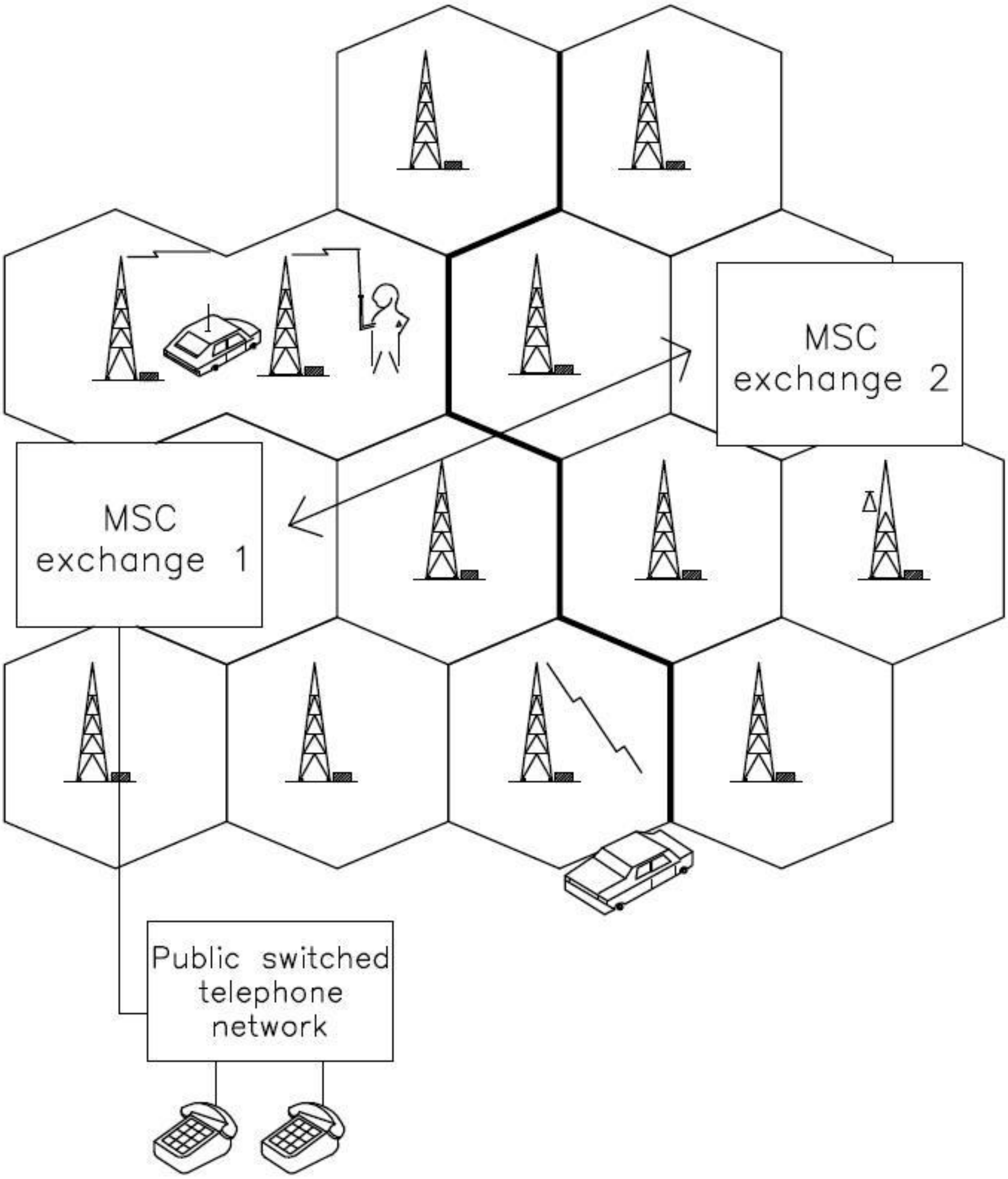


Figure 2: Mobile switching centres

The TACS system has subsequently become known as the first generation of mobile telephony.

2.3.2 Mobile activity

On joining the network, mobiles must first register their presence with the PLMN, but to do this they must find out the parameters of the cell they are in. This in turn requires that base stations transmit a continuous “forward control channel” (FCC), which mobiles search for on allocated control frequencies. Mobiles can respond on a “reverse control channel” (RCC). Since TACS is an analogue system, the channel coding available for protection of both signalling and voice data is limited and in control channels transmissions rely on repetition of control words to ensure reliability of reception [11].

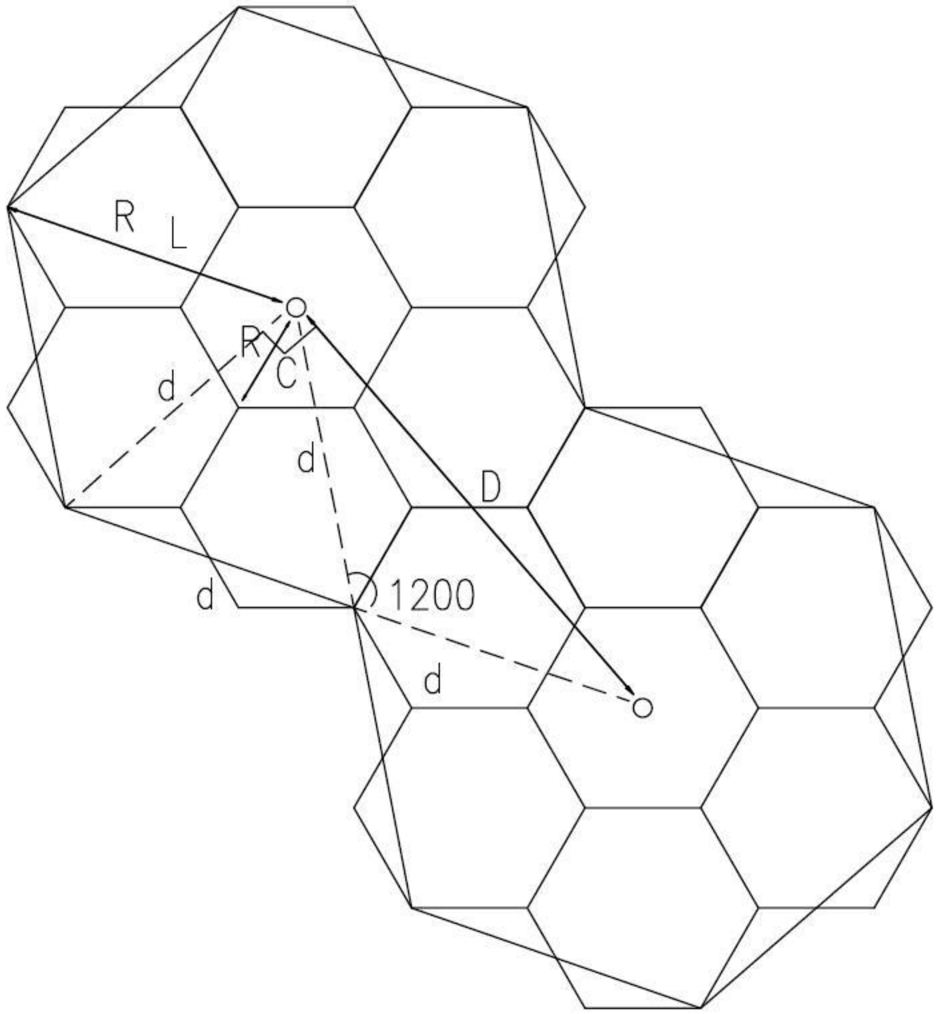
The mobile also receives instructions from the base station about what transmit level to use and it also notes, via a “busy-idle” bit stream on the FCC, when another mobile is up-linking on the RCC to the base station to avoid contentions.

2.3.3 Hand-over

It is worth noting the hand-over procedure in TACS in order to draw comparisons with the equivalent process in GSM and UMTS. In TACS, hand-over is controlled entirely by the MSC which is handling a call that needs to be transferred to a new cell. When the MSC detects that the received up-link signal is falling due to the mobile’s progress away from the base station, it initiates a search for an available channel in one of the surrounding cells. If a free channel is available, the mobile is instructed to pick it up so that the call can continue. In other words, the mobile is an entirely passive participant in the hand-over process.

2.3.4 Limitations of TACS

The radio spectrum allocated to TACS was limited (872MHz to 965MHz) since the system had to work in duplex mode. It is not hard to see that with 25kHz allocated to each paired frequency the system capacity is extremely constrained. In order to achieve acceptable speech quality in each channel and control the co-channel interference problem a frequency reuse pattern corresponding to a seven-cell cluster was needed.



$$\frac{R_L}{R_L} = \sqrt{N}$$

$$D = \sqrt{(2d)^2 - d^2}$$

$$= \sqrt{3}d$$

But $d = R_L$

$$\therefore D = \sqrt{3} \cdot \sqrt{N} R_C$$

$$\therefore \frac{D}{R_C} = \sqrt{3N}$$

N = number of cells in each cluster i.e. 7

Figure 3: The limitations of analogue technology

This figure and associated calculation clearly demonstrate the severe limitations of analogue technology. The objective in subsequent developments in digital system technology had to aim to at least a four-cell reuse pattern and, ultimately to the reuse all the available spectrum in every cell.

2.4 The GSM system

2.4.1 Functionality

Although GSM originally stood for *Groupe Special Mobile* – the European secretariat set up in 1982 to create a common European mobile telephone standard that would formulate specifications for mobile cellular radio system operating at 900MHz – it was subsequently used to refer to the "global system for mobile" communication, reflecting the fact that the application of GSM would span much further than Europe alone. While GSM includes detailed specifications of functions and interface requirements, it does not deal with the hardware. The aim, then, was to enable operators to buy equipment from different suppliers while limiting designers as little as possible.

The first important decision, taken after field testing, was to favour a digital system over analogue for GSM. The second was to adopt the narrowband time division multiple access (TDMA) rather than broadband. This allowed many channels to be supported by a single base station transceiver. However, if the number of channels for each carrier – and so the bit rate – is too high, it could create difficulties for mobiles in having time to scan a number of base station transmissions to determine hand-off requirements [4].

GSM has a harmonized spectrum, which means that even though different countries may operate on different frequency bands, users can transfer seamlessly between networks and keep the same number. As a result, GSM users have widespread coverage across the planet. And because GSM is used throughout the world, there is a significant variety of telephones that operate on GSM. Consumers are therefore not limited to buying telephones made in their respective country and can select, from a wide variety, a handset that fits their specific needs. Because GSM is the same network worldwide, users are not charged a roaming fee for international calls – though most providers still charge a service fee on international calls [12].

GSM has become known as the second generation of mobile telephony. The GSM network incorporates: the switching system (SS); the base station system (BSS); and the operation and support system (OSS) [13].

2.4.1.1 The switching system

The switching system (SS) is responsible for performing call processing and subscriber-related functions. The switching system includes:

- home location register (HLR) — the database of subscribers permanently registered in the system administered by a particular operator;
- mobile services switching centre (MSC) — which performs the telephony switching functions of the system;
- visitor location register (VLR) — the database containing temporary information about subscribers that is needed by the MSC in order to service visiting subscribers;
- authentication centre (AUC) — the database that allows verification of a user's identity and ensures the confidentiality of a call; and
- equipment identity register (EIR) —the database of serial numbers of the mobile phones used in a system, which helps to prevent calls from stolen, unauthorised or defective mobiles.

2.4.1.2 The base station system (BSS)

The base station system (BSS) consists of base station controllers (BSCs) and the base transceiver stations (BTSs). The BSC is a high capacity switch that provides functions such as hand-over, cell configuration data and control of radio frequency (RF) power levels in base transceiver stations. The BTS is the radio equipment needed to service each cell in the network. A group of BTSs is controlled by a BSC.

2.4.1.3 The operation and support system

The operations and maintenance centre (OMC) is connected to all equipment in the switching system and to the BSC. It performs such management functions as tariff accounting, traffic monitoring and management in case of failures of particular network blocks. An important task of the OMC is management of the home location register (HLR).

2.4.2 Mobile activity

When the mobile phone is switched on, it “finds itself” in the GSM network. This involves: looking for the carrier on which, in the local cell, the broadcast channel is transmitted; listening to carriers listed according to their decreasing power, to search for the frequency correction channel; adjusting the mobile phone carrier frequency generator to that frequency; finding other important control channels; locating the synchronisation channel and decoding the information it contains; and decoding the information carried by the broadcast control channel.

Now the mobile phone has to be registered with the network. Registration occurs if the number received from the base station differs from the latest one that has been stored in the mobile phone. This is followed by the authentification procedure. Only when the mobile phone has been fully synchronised with the network and registered can the call set-up finally begin.

The mobile phone listens to the common control channel in order to find a paging channel directed to it. After receiving the access granted channel, the stand alone control channel is assigned to the mobile phone. Using that channel, the mobile tells the base station controller about the call set-up request. The BSC transfers that message to the mobile services switching centre (MSC), which informs the visitor's location register (VLR) associated with it about the call set-up request. An identity number is now transmitted to the mobile phone, which goes on to send to the MSC and VLR the number called and service requested. The number is now called. When the recipient accepts the call, voice or data transmission between the two is possible [13].

2.4.3 Hand-over

Particular attention was paid to hand-over when the GSM system was developed.

The intra-cell hand-over is the easiest type. It takes place if it is necessary to change the frequency or slot being used by a mobile because of interference or for other reasons. The traffic load in the cell can be optimised or the quality of a connection improved by changing the carrier frequency.

An intra-BSC hand-over occurs when the mobile moves out of the coverage area of one BTS and into another controlled by the same BSC. The BSC assigns a new channel and slot to the mobile, then releases the old BTS from communicating with the device.

An inter-BSC hand-over takes place when the mobile moves out of the range of cells controlled by one BSC. The hand-over is then from one BTS to another and one BSC to another. This type of hand-over is controlled by the MSC.

An inter-MSC hand-over occurs when changing between networks. The two MSCs involved – which can be located more than 100km from each other – “negotiate” to control the hand-over. This type of hand-over makes especially high requirements of the cellular network.

In first generation analogue systems, hand-over is initiated on the basis of the quality or field strength measurements in the up-link made by the network. This puts a considerable load on the network. In the GSM system, the measurements are performed both in down-link and up-link, so the mobile station takes part in the hand-over. Both transmission directions are used to measure the quality and level of the received signals.

The mobile station regularly measures the 16 strongest carriers transmitting the broadcast control channel. The measurements of the six strongest carriers are transmitted to the currently assigned base station. The base stations also measure interference in any free time slots. The operation and maintenance centre, which supervises the operation of particular GSM system blocks, monitors cell traffic levels, which can also be used in the hand-over procedure [14].

2.4.4 Limitations of GSM

A significant drawback of GSM is that multiple users share the same bandwidth. Interference on the transmission can occur when there are enough users. Faster technologies, such as 3G, have been developed on different types of networks from GSM, such as CDMA, to avoid this problem.

Another disadvantage of GSM is that, since it uses a pulse transmission technology, it can interfere with electronic devices like pace makers and hearing aids.

As a result, cell phones often have to be turned off in such places as intensive care units and aircraft [12].

2.5 The 3G system

2.5.1 Functionality

As mobile phones became almost ubiquitous, users began to demand a wider range of services from them – most obviously internet access. Second generation technology was inadequate to supply this effectively, so developers began to work on ways of providing much higher data transmission rates and offering increased capacity.

The main difference between 2G and 3G technology is that while the former uses time division multiple access (TDMA) as its transmission system, the latter is based on code division multiple access (CDMA) technology. CDMA makes better use of the available spectrum because it allows all base stations to use the same frequency. Instead of circuit switching, which forms the basis of 2G technology, in 3G systems data is split into separate packets and transmitted, before being reassembled in the correct sequence at the receiver end using a code that is sent with each packet. This code depends on an accurate time stamp being put on each piece of the signal, and 3G depends on the global positioning system (GPS) for this information.

CDMA digitises all data – including voice signals – and spreads it over the entire available bandwidth. This enables multiple calls to be overlaid on each other on the channel. Between eight and 10 separate calls can be carried in the same channel space as one analogue AMPS (advanced mobile phone system) call [15].

Third generation systems usually use parts of the radio spectrum around 2GHz, well away from the crowded frequency bands used in 2G technology. Data rates vary from 144kbps for moving vehicles to 384kbps for pedestrians and up to 2Mbps for indoor or stationary users. These figures contrast with the 9.6kbps supported by basic 2G networks [16].

Available on 3G handsets are TV streaming, multi-media, videoconferencing, web browsing, e-mail, paging, faxing and navigational maps.

About 10 per cent of the estimated five billion mobile phones on the planet are 3G. There are, as yet, only two million subscribers to the next generation, the 4G networks, scattered across the USA and Far East [17].

2.5.2 Mobile activity

The two main 3G systems are UMTS (universal mobile telecommunications system) and cdma2000 (code division multiple access 2000).

UMTS was developed mainly for countries with GSM networks, because these countries agreed to free new frequency ranges for UMTS networks. An entirely new radio access network had to be built for UMTS, because it is a new technology and in a new frequency band. The advantage of this is that the new frequency range gives plenty of new capacity for operators.

UMTS is a real global system, comprising both terrestrial and satellite components. It offers a consistent service environment even when roaming via VHE (virtual home environment). A person roaming from his or her network to other UMTS operators experiences a consistent set of service, and so the feeling of being on his or her home network, independently of the location or whether access is terrestrial of via satellite [18].

The cdma2000 specification was developed by the Third Generation Partnership Project 2, consisting of telecommunications standards bodies from Japan, China, Korea and north America. Cdma2000 was developed to be compatible with cdmaOne which, also known as IS-95, was the reference system in the early 1990s because of the innovative solutions it applied. This compatibility was needed for successful deployment for US market. Cdma2000 is easy to implement because operators do not need new frequencies [18].

With cdma2000, the whole spectrum can be reused in each cell. This, though, can lead to interference among users. Indeed, the main limitation of cdma2000 is the interference generated by users with respect to each other. Keeping the signal power at the lowest level compatible with reception quality helps to minimise this. Cdma2000 also requires precise power control of user signals, to avoid the near-far effect – a condition in which a strong signal captures a receiver making it impossible for the receiver to detect a weaker signal [19].

There has been some harmonisation between the UMTS and cdma2000 systems in areas like chip rate and pilot issues.

2.5.3 Hand-over

The main reason for performing hand-overs is that a user has moved and he or she can be served in another cell more efficiently, perhaps with less interference or using less power. But hand-overs may also be performed for other reasons, such as system load control. Hand-overs can be from a 3G system to a 2G system (and *vice-versa*) or from one 3G system to another.

Hand-overs in 3G systems can be hard or soft. With hard hand-overs, all the old radio links in the user equipment are removed before the new radio links are established. Hard hand-overs tend to be used when a change of the carrier frequency is needed.

With soft hand-overs, the radio links are added and removed in such a way that the user equipment always keeps at least one radio link to the terrestrial radio access network. Soft hand-overs tend to be used when cells operated on the same frequency are changed [20].

2.5.4 Limitations of the 3G system

The main limitation of 3G centres on available bandwidth. 3G networks have a maximum bandwidth of 2Mbps. Realistically, though, actual bandwidth is often around 384kbps. This, of course, is much better than the bandwidth available in 2G systems, but it is not really enough for effective multi-media communication.

A second limitation is that 3G has failed to achieve the goal of global roaming. The International Telecommunication Union envisaged a single radio interface to provide global roaming. In the end, though, five radio interfaces were adopted for 3G networks to cater for competition and migration of the installed base of 2G networks.

Thirdly, 3G specifications define three different core network domains, each providing a different set of services. The CS domain provides circuit switched services, the PS domain provides packet switched services and the IMT domain provides IP multi-media services. This is inefficient and limits the scope for development.

Fourthly, the sluggish pace of 3G network deployment enabled other wireless technologies – such as wireless LAN (WLAN) and Bluetooth – to capture quite a bit of the market.

Finally, the amount of "chatter" between the handset and the network when the user tries to do something using 3G technology is inefficient and can lower the quality of the service on offer.

2.6 Fourth generation networks

4G has been developed as a successor to 3G networks, not only to overcome the limitations of 3G but also to make use of the latest developments in wireless technology.

The main difference between 4G and earlier systems is that 4G uses orthogonal frequency-division multiplexing (OFDM) rather than time division multiple access (TDMA) or code division multiple access (CDMA). OFDM is a type of digital modulation in which a signal is split into several narrowband channels at different frequencies. This is more efficient than TDMA, which divides channels into time slots and has multiple users take turns transmitting bursts, or CDMA, which simultaneously transmits multiple signals on the same channel.

OFDM allows for the transfer of more data than other forms of multiplexing. It simplifies the design of the transmitter and receiver and allows for use of almost the entire frequency band. No gaps are needed to prevent interference. The frequencies are spaced so that the signals do not interfere with each other and so there is no cross talk. Parallel data transmission allows multiple signals to be sent simultaneously from the same antenna (or wire) to one device. Each transmission has a different stream of bits [21].

According to the International Telecommunication Union, a 4G network requires a mobile device to be able to exchange data at 100Mbit/sec, while a 3G network can offer data speeds as low as 3.84Mbit/sec [22].

Research commissioned by UK telecommunications watchdog Ofcom indicates that early 4G mobile networks with standard configurations will be 3.3 times more spectrally efficient than standard 3G networks [23].

It is expected that 4G will be able to offer so-called “pervasive computing”, in which simultaneous connections to multiple high-speed networks will provide seamless hand-overs throughout a geographical area. New coverage enhancement technologies are being developed to address the needs of mobile users in homes and offices. These will liberate network resources for users who are roaming or in more remote areas [22].

CHAPTER 3

Strategies for supporting quality of service

3.1 General features

The extensive literature referred to in Chapter 1 dealing with the complexity of the mobile radio propagation channel from fixed base stations to mobiles and *vice-versa* clearly demonstrates the challenge faced by system designers. Apart from the general decline in signal strength with distance of mobiles from base stations, the effects of shadowing by obstacles and multi-path propagation resulting from reflections and refractions create rapid fading of received signals. Fades can be deep enough to effectively prevent communication, but since fades are short-lived means can be found to overcome this feature of the channel.

A further consideration is the different demands of speech and data, which are manifested in the strategies devised for managing transmissions. In general terms, voice transmissions will tolerate minimum delay but some degradation of speech quality is acceptable provided the speech is intelligible. Data, on the other hand, can tolerate delay but transmission errors are not acceptable. In either case there is clearly a trade-off between through-put and integrity of transmitted traffic. These aspects are illustrated in the strategies adopted in the various generations of systems.

3.2 First generation systems

The first generations systems, as commented earlier (AMPS in USA and TACS in UK) used analogue frequency modulation operating single channel per carrier. Since there are no signal processing options in analogue transmissions, the integrity of control signalling is the most important aspect. The allocated spectrum is divided into control channels and voice channels with different FM peak deviations set to take account of the differing spectral properties of voice and data.

In the control channels each control word (base to mobile) is repeated typically six times so that a mobile could receive all the repetitions and compare them bit by bit to get the best estimate of the transmitted word. The estimate usually has one or two residual errors and also check bits, so that by using the properties of a BCH (Bose, Chaudhri, Hocquenghen) code up to two remaining errors can be corrected [4].

A modest protection from co-channel interference for voice transmissions is provided by a supervisory audio tone (SAT) transmitted continuously during a call. The SAT is one of three frequencies above the receiver audio response allocated on a three cell cluster basis. A mobile receiving a call notes the SAT frequency it is allocated and, if a strong co-channel interferer attempts to capture the receiver, its unwanted SAT is recognized and the receiver mutes its audio output until the interference subsides.

3.3. Second generation systems

The most successful second generation system is the GSM system, referred to in Chapter 2. This section describes the strategies developed in GSM to combat the variability in the received signal.

3.3.1 Speech coding

Unlike the waveform digitization of speech in fixed networks, the GSM speech coder uses a process of analyzing speech into the constituent parts of excitation and vocalization. The excitation part repeats on a regular pulse pattern (RPP) while the vocalization element is transmitted effectively as the parameters of a filter which characterizes the vocal tract. In addition the GSM coder incorporates a long term predictor (LTP). The process is presented in Figure 4, in which LPC is linear predictive coding and RPE is regular pulse excitation.

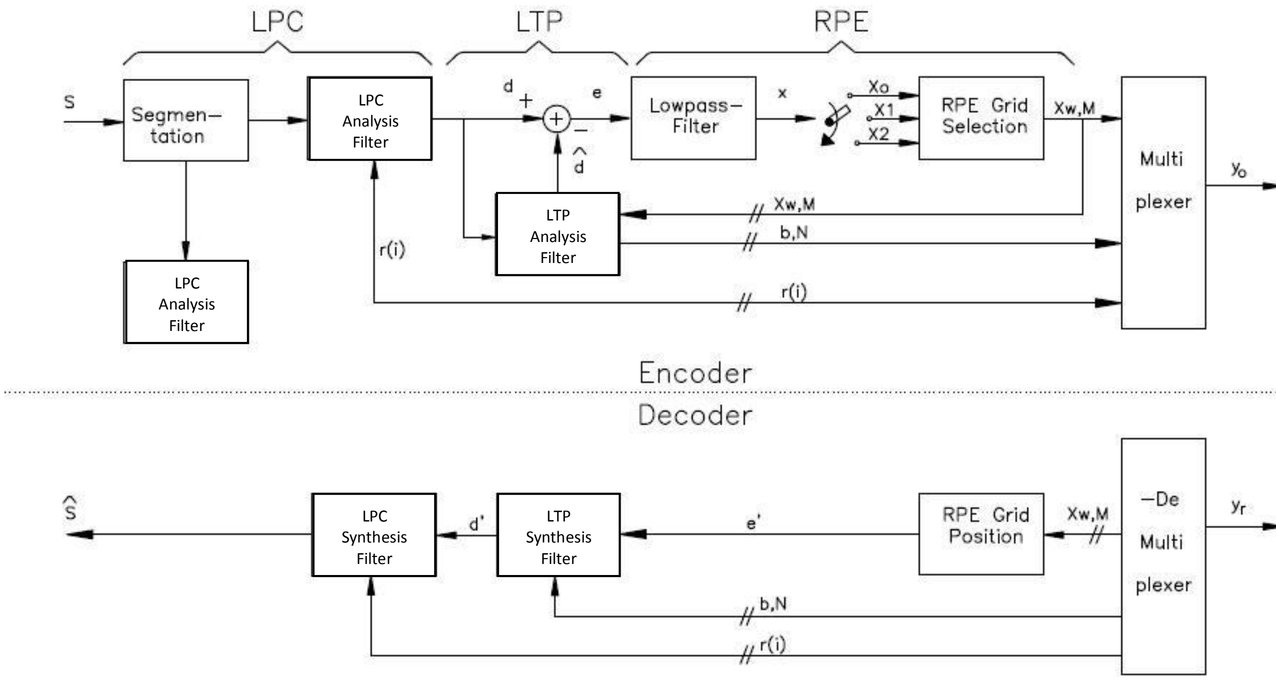


Figure 4: Block diagram of GSM speech coder [24]

The coder outputs at the rate of 13kbits per second and samples of the input speech are taken every 20 milliseconds giving an output of 260 bits per sample.

As speech coder developments have progressed it has become possible to obtain satisfactory coder output at half the full rate. The output bit stream from either full rate or half rate coders has most significant bits and least significant bits with a substantial number of bits which do not fall into either category.

3.3.2 Coding and interleaving

The question then arises as to how to deliver the 260 bits from each coder sample over the radio channel. There are several possibilities but GSM uses two in combination. One is to add a substantial overhead to correct for bit errors; the second is to disperse each sample over a number of TDMA frames so that if a frame is lost because of fast fading much of the speech sample is retained. The process is illustrated in Figure 5.

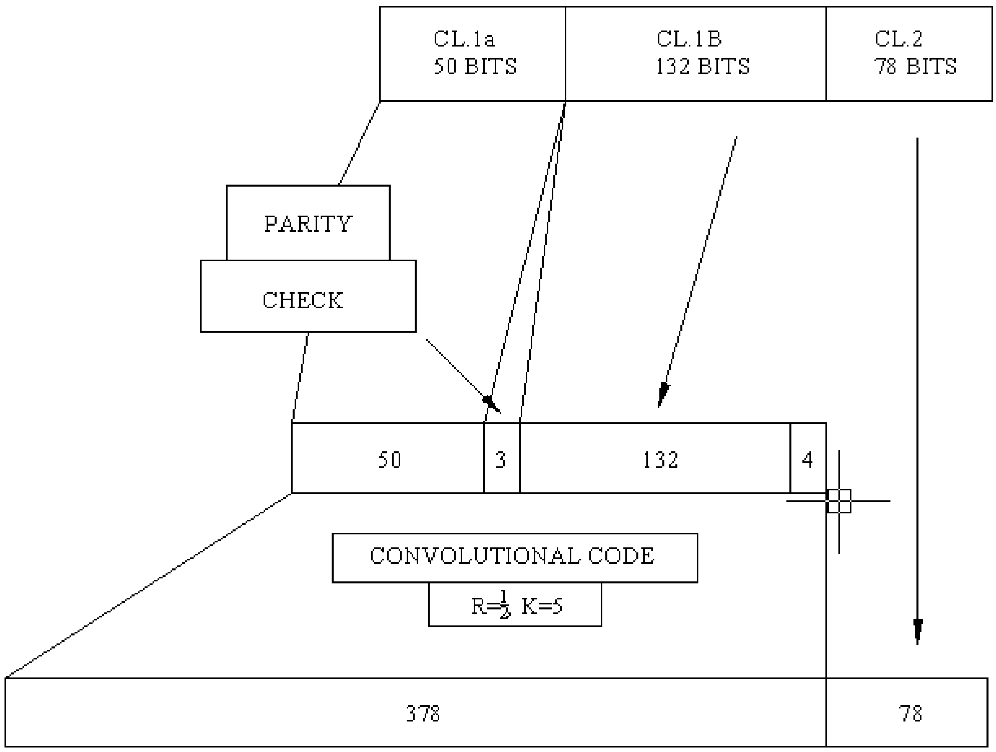


Figure 5: Protection of full rate speech

- Class 1a 50 bits – most sensitive to bit errors
- Class 1b 132 bits – moderately sensitive to bit errors
- Class II 78 bits – least sensitive to bit errors.

Class 1a bits have a three-bit cyclic redundancy code added for error detection. If an error is detected, the frame is judged too damaged to be comprehensible and it is discarded. It is replaced by a slightly attenuated version of the previous correctly received frame.

The 50 most significant, or Class 1a bits, are protected by a 3 bit parity check and then combined with the Class 1b 132 bits which themselves are followed by 4 bit tail sequence. The selection of which coder output bits are assigned to which category was determined by an extensive campaign of subjective testing. These 189 bits are then encoded using a half-rate convolutional code with constraint length 5. Effectively this encodes each input bit into 2 output bits yielding 378 bits. Finally, the remaining 78 bits from the 260 are added without protection to inflate the original 20 millisecond sample to 456 bits.

Interleaving of speech frames is then accomplished by sub-dividing the 456 bits into eight 57 bit blocks which are dispersed over eight TDMA frames, as shown in Figure 6.

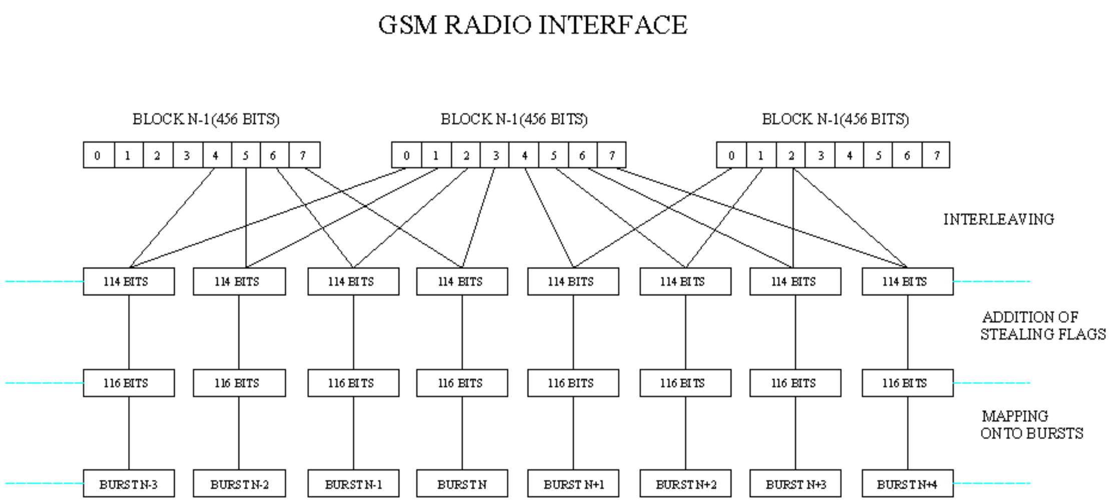


Figure 6: Interleaving speech frames on to TDMA frames

Recalling that the TDMA frame structure involves eight time slots per frame, this means that each time slot carries two 57 bit sub-blocks drawn from two different speech samples.

There is a further means to combat the effects of the radio channel. This involves equalizing the channel and allowing receivers to compensate for the rapidly changing channel characteristics. This is accomplished by inserting a training sequence between the two 57 bit sub-blocks in each time slot. The receiver then makes adjustments to its equalizer to obtain the best match to the training sequence and then uses this equalizer setting until the next training sequence is detected. This gives the final total bits per time slot as 148 bits.

3.3.3 Duplex operation and hand-over

In first generation analogue systems (TACS/AMPS) achieving full duplex operation required a high specification duplex filter to separate up-link and down-link paths when active simultaneously. In GSM this is avoided by off-setting the mobile receive time-slots from the up-link transmit slots by three slot intervals. This gives mobiles time to change from transmit to receive so that up and down-link activity does not happen simultaneously.

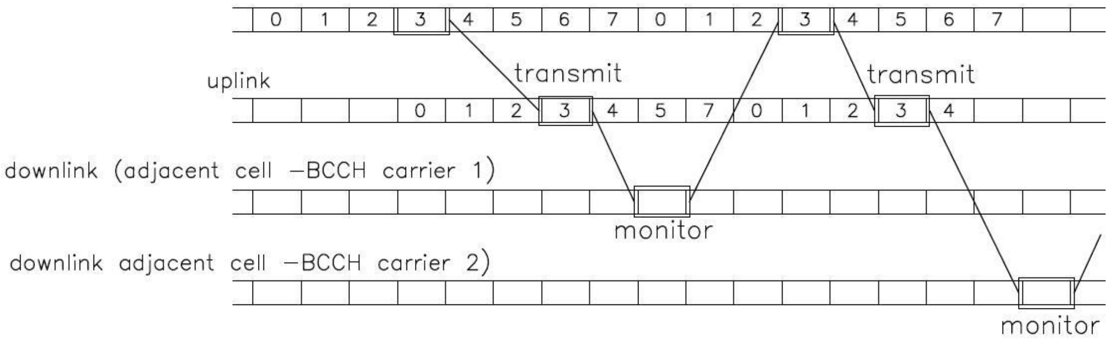


Figure 7: Offset of up-link and down-link traffic

The hand-over process also differs in GSM from TACS/AMPS. In GSM, during a call, mobiles are active for two out of the eight time slots available, one down-link and one up-link. This means that mobiles can use their vacant time slots to listen to down-link transmissions from base stations in surrounding cells. Mobiles can then compare received signal strengths from adjacent cells and determine when a hand-over would be appropriate.

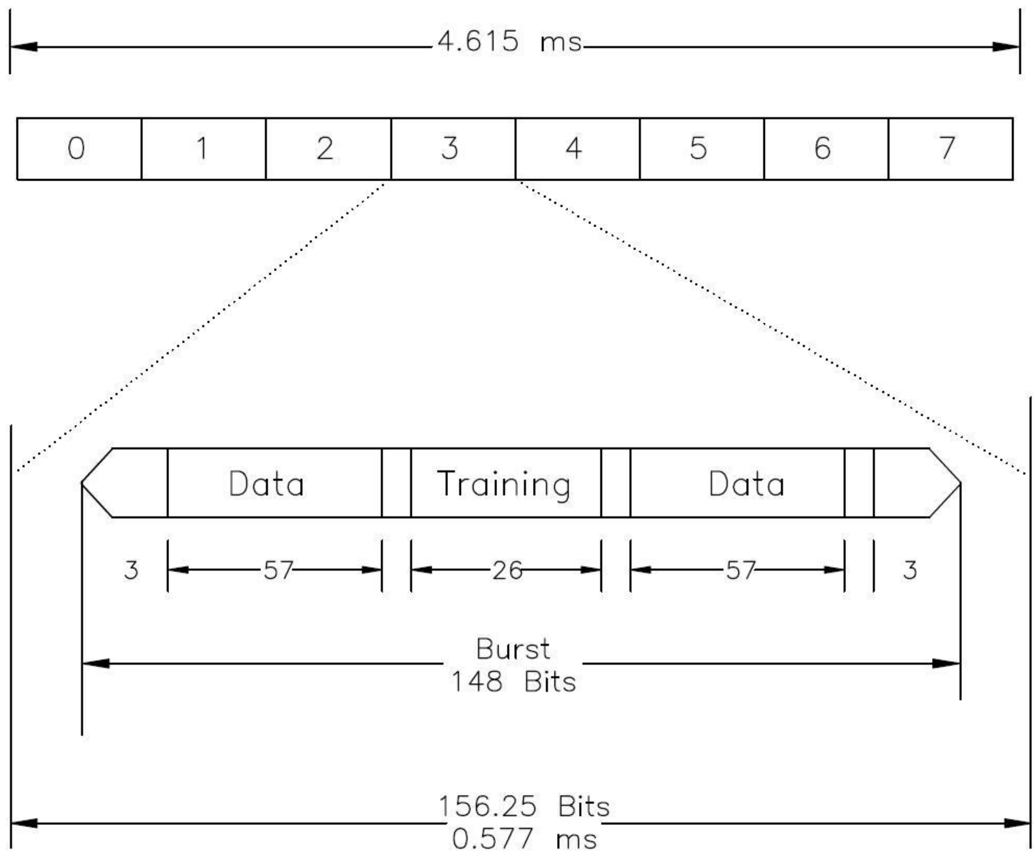


Figure 8: Basic time frame, time slot and burst structures

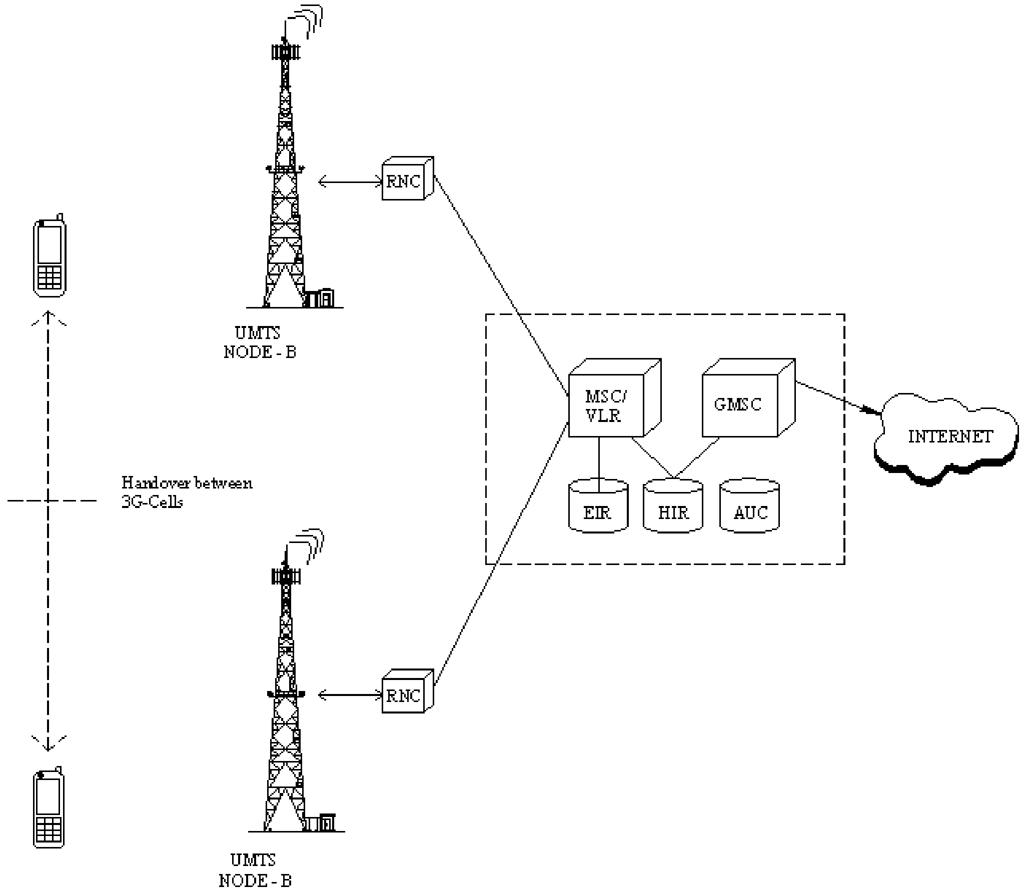


Figure 9: Mobile terminal hand-over support [25]

3.3.4 Data services

As noted previously, data services cannot tolerate the bit error rates which are acceptable for speech. Data protection therefore requires that throughput is sacrificed in order to maintain the integrity of the data connection. This results in a data transfer rate in GSM of 9.6kbps.

3.4 Third generation systems

The first and second generation systems have used a channelized approach to enabling users to access radio resources, single channel per carrier in 1G, eight users per carrier in 2G GSM. In 3G a fundamentally different approach is used; the spread spectrum process by which multiple users share the same spectrum simultaneously. Individual communications between base stations and mobiles are distinguished from all other users by assigning an individual a unique code to each individual transmission. Only the intended recipient can know the unique code so that applying a correlation process to the total received signals from all users results in extraction of only the wanted messages. Equally, all other received signals do not result in a correlated output and appear only as noise. The history of code division multiple access (CDMA) began in military communications in the 1960s but it was only in the 1990s, thanks to advances in integrated circuit design and realization in silicon, that the technology became viable for cellular communications.

3.4.1 The IS-95 system

The IS-95 system was developed in USA by the Qualcom company beginning in 1995. By 1997 the first IS-95 products became available commercially and were intended to be an alternative to GSM – i.e. IS-95 was a second generation system. It was, however, the first to use CDMA in cellular radio applications. The principal radio frequency parameters were:

- carrier spacing: 1.25MHz

- chip rate: 1.2288Mcps
- power control: down-link, slow; up-link 800Hz

Other features were that base stations need to operate synchronously using GPS (global positioning system) timing signals from the GPS satellite constellation. Also, data were transmitted as short circuit-switched calls. While IS-95 was subsequently developed into cdma2000, it remained an essentially US-based system. A further version, IMT2000, targeted international adoption, but in Europe the principal system for 3G was universal mobile telecommunications system (UMTS).

3.4.2 UMTS

The CDMA technology offers much greater flexibility in supporting differing demands and services in the air-interface.

3.4.2.1 RF parameters

- Carrier spacing: 5MHz
- chip rate: 3.84Mcps
- power control: 1500Hz up-link and down-link
- packet data: load-based scheduling

Additionally, UMTS base stations operate asynchronously and support inter-frequency hand-overs.

3.4.2.2 Quality of services (QoS) classes

In UMTS four classes of traffic have been specified: conversational; streaming; interactive; and background. The conversational class is intended for very delay sensitive applications such as voice, video-telephony and video games; the streaming class is for streaming multi-media; the interactive class targets web browsing and network games; the background category is for e-mails where delay in delivery is not significant.

3.4.2.3 Support for speech services

Speech services fall into the conversational class but within the class further possibilities must be taken into account which will affect speech quality. UMTS employs an adaptive multi-rate (AMR) technique. This uses a single integrated speech coder capable of delivering a range of output bit-rates:

> 12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 515 and 4.75kbps. The 12.2kbps output corresponds to the GSM coder, other bit-rates correspond to IS-641 (for IMT2000) etc.

In UMTS the speech sampling interval remains 20 milliseconds as in GSM but the speech coder output can be changed every 20 milliseconds to accommodate changes in loading on the system and the consequent need to reduce coder output bit-rates when loading is high and more users need to be supported.

On the other hand, speech coder technology has advanced since the inception of GSM and the UMTS coder uses an algebraic code excited linear predictive coder (ACELP) so that the lower bit-rate outputs can support speech transmission with only small degradation of speech quality. Nevertheless, these considerations have significance in relation to subsequent discussion of call charging strategies related to speech quality since the process of moving among the speech coder output rates is under the control of the network operator.

3.4.2.4 The UMTS physical layer

Fundamental aspects distinguish the UMTS CDMA capability from previous generations of system. Of primary significance is the consequence of spreading the input base-band signal to occupy the spread bandwidth of 5MHz. In this bandwidth the coded signal appears as 3.84MHz. The ratio of spread bandwidth to base bandwidth is the spreading gain. Taking the 12.2kbps output from the speech coder the spreading gain is:

$$10 \text{ Log } (3.84e6/12.2e3) = 25\text{dB}$$

After de-spreading the receiver will need to achieve around 5dB carrier to noise ratio. Therefore the received signal can be – 20dB below the ambient noise. By comparison, GSM requires around 9-12dB carrier to noise.

Additionally, various levels of modulation can be accommodated and different coding schemes to support different data integrity requirements. The system approach is fully set out in [26]. However, this flexibility comes at the price of greatly increased system complexity.

3.2.4.5 The near-far problem

Separating the simultaneous user equipment (UE) transmissions at the base station uses a correlation procedure which is only effective if the received signals from mobile terminals arrive at approximately the same field strength. Therefore, UEs near to the base station must be instructed to transmit at a low level while distant UEs must increase their output. This would not be a problem if the radio connection were stable but, as noted above, the radio channel is characterized by fast fading so the transmissions from mobiles must be adjusted at the fading rate. Figure 10 [26] illustrates this process, with mobile terminals adjusting their transmissions at 1.5kHz or every 0.67msec.

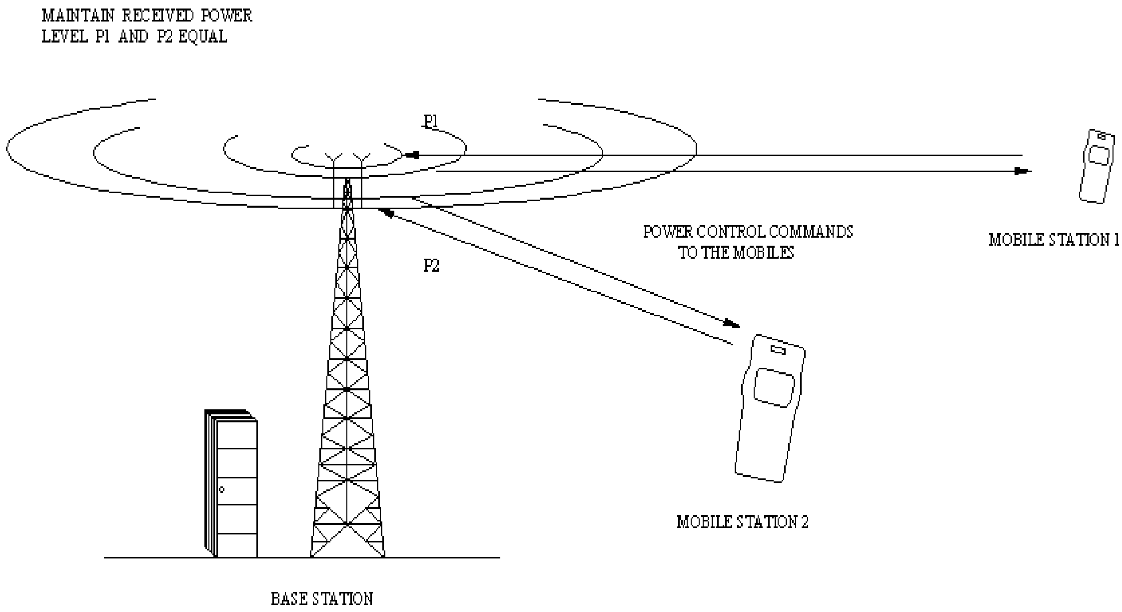


Figure 10: Closed loop power control in CDMA

3.4.2.6 Rake reception

Signal variability has been commented on above. However, there are aspects of multi-path propagation which can be used to advantage in UMTS receivers. In Figure 11 [26] the different propagation paths from base to mobile are seen to result in the multi-path signals arriving at different times. Bearing in mind that the process of recovering received information is one of correlation, it follows that if several versions of the same message arrive dispersed in time each version can be correlated sequentially. This means that a correlator must be assigned to each received version, giving a process akin to the tines of a rake collecting signals. Hence the term rake reception.

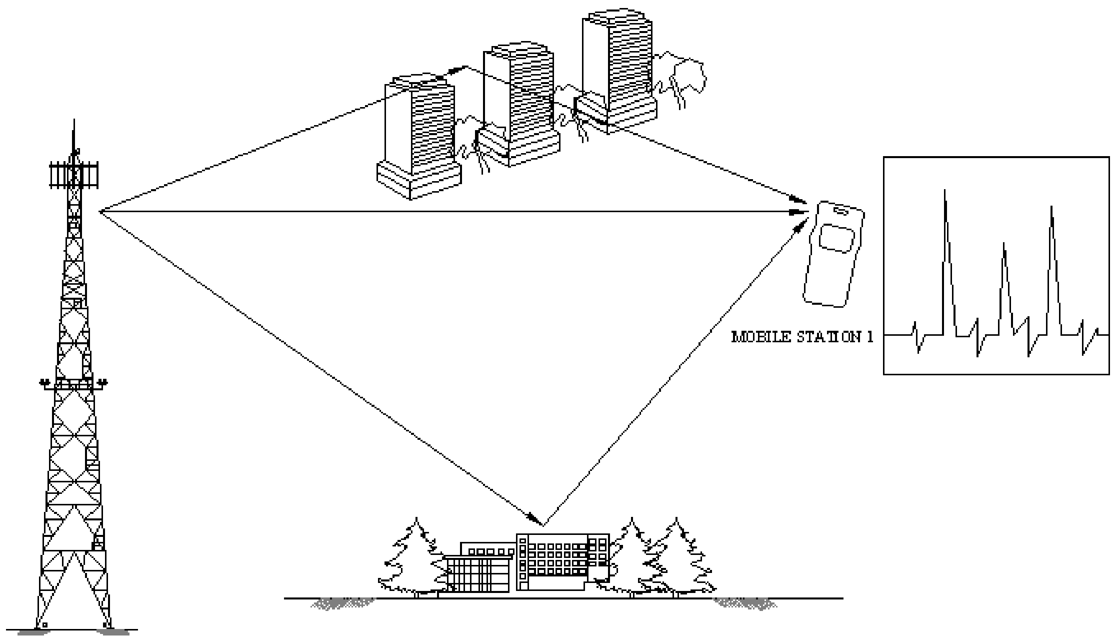


Figure 11: Multi-path propagation leads to multi-path delay profile

Of course, the received signals from the three multi-paths do not align in-phase, so some operation is required based on estimation of the radio channel. Then maximal ratio combining with the CDMA rake receiver can be achieved as shown in Figure 12 [26].

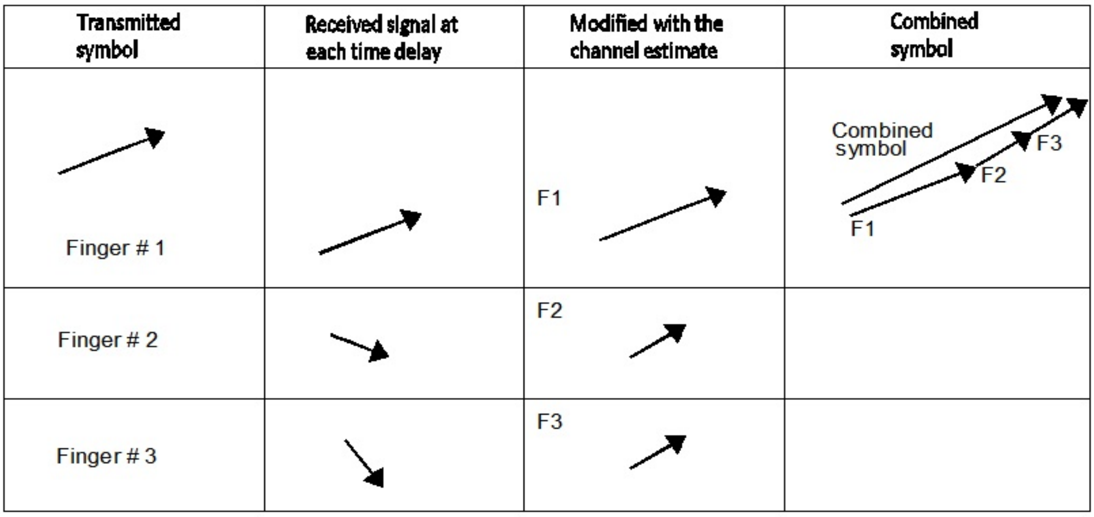


Figure 12: Principle of maximal ratio combining with the CDMA rake receiver

The receiver functionality is presented in Figure 13 [27] where the processes of channel estimation and phase rotation are seen in relation to the code generation and correlation functions.

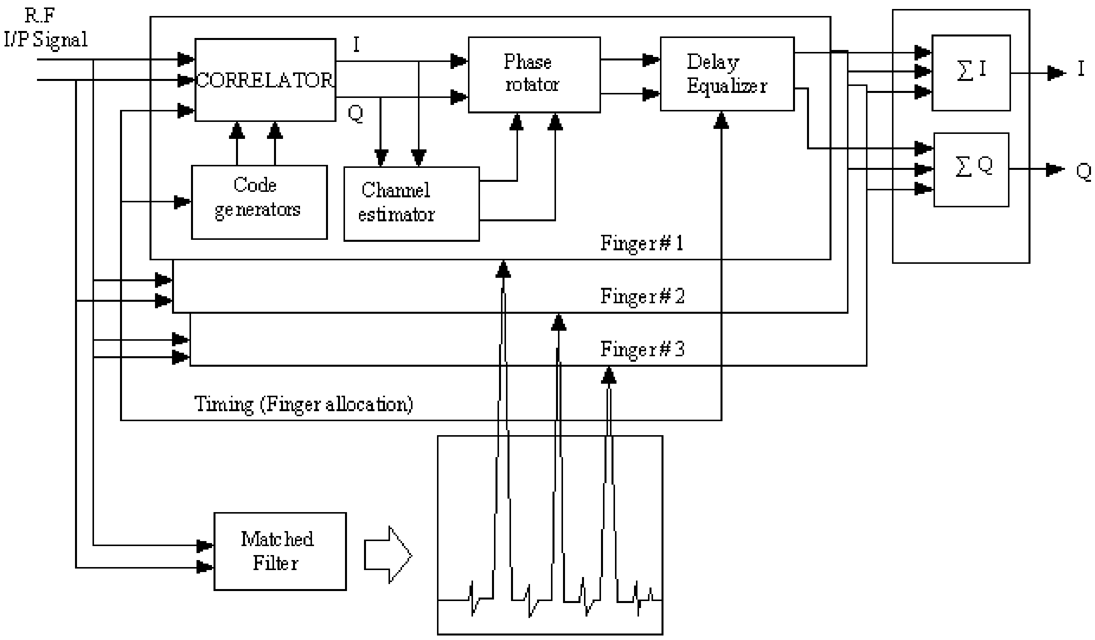


Figure 13: Block diagram of CDMA rake receiver. This performs the phase alignment of Figure 12

3.4.2.7 Hand-over

As with other aspects of CDMA systems technology, hand-over processes can be implemented in two ways, by “soft” and “softer” hand-over. These are illustrated in Figures 14 and 15 [26].

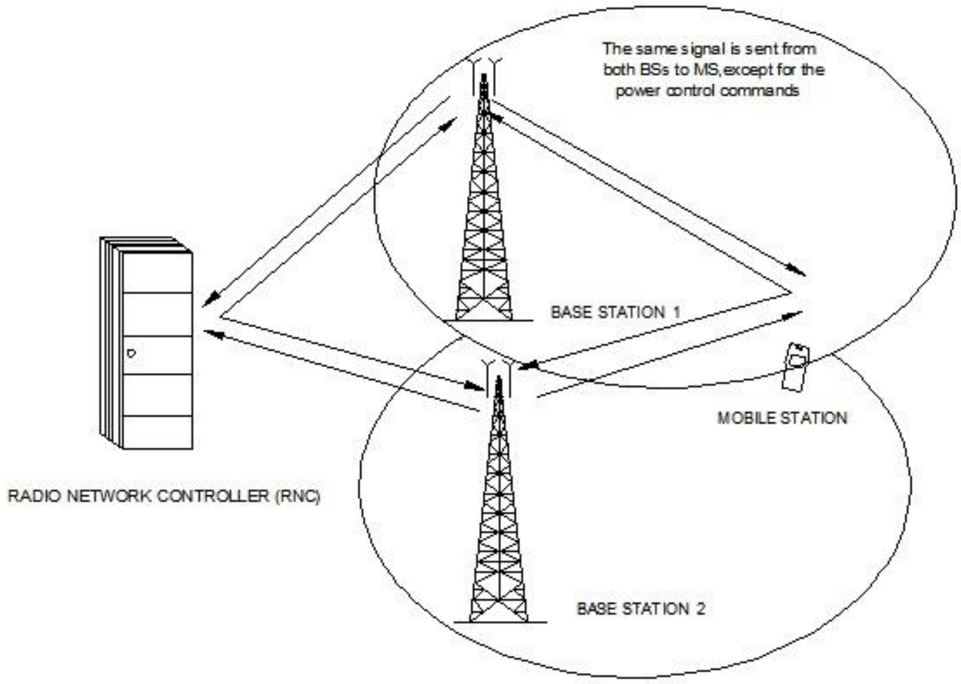


Figure 14: Soft hand-over. In soft hand-over, the mobile is moving from the coverage area of one base station to the coverage area of a second base station. Both base stations maintain the call until the transmission is complete.

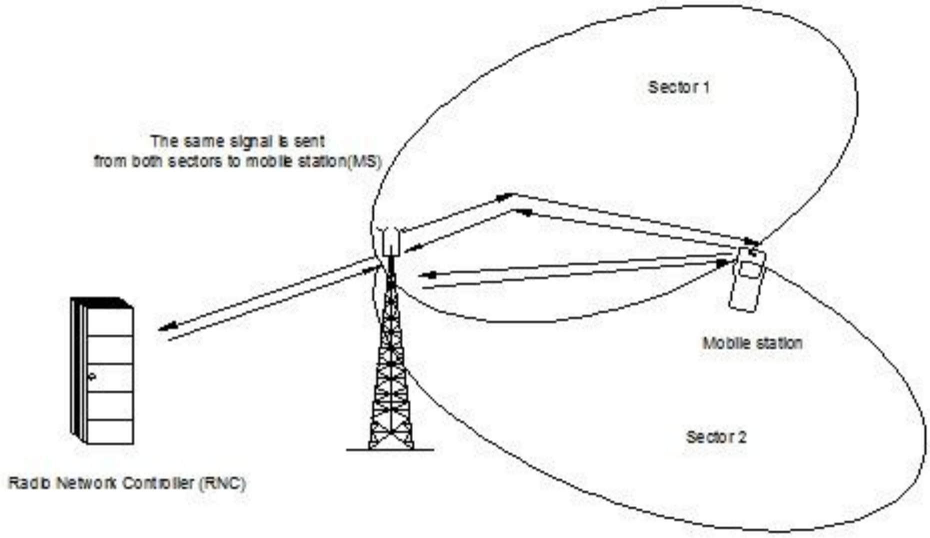


Figure 15: Softer hand-over. In softer hand-over, the same base station maintains the call as the mobile transits from one segment to another without the need to change to another base station.

These operations are possible since base stations in adjoining cells can engage a UE before the base station that has supported it has to disconnect.

Backwards compatibility with GSM was an important consideration in the development of UMTS. Hand-over from UMTS to GSM and *vice-versa* is assisted in the soft and softer hand-over functions in UMTS. These features also facilitate inter-system hand-overs from time-division duplex operation.

In soft hand-over, the mobile is moving from a cell supported by one base station to a cell supported by a second base station. The mobile gains access to the new cell before releasing the old one. In softer hand-over, mobiles moving from one cell segment to another supported by the same transmitting site can again pick up the new segment before releasing the old one.

3.5 Other related system considerations

Returning to the user classes referred to above, it is appropriate to consider both other types of service and other elements of complete systems.

3.5.1 Progress to multi-media services

Multi-media transmission to mobile terminals presents a significantly demanding challenge to network designers and to advanced mobile terminal capability. Two approaches are the basis of this service provision. Source material generally contains redundancy and this can be removed. Also, there are well established methods for compression of digital source material which can be decompressed at the receiving end. This compressed material is then the suitably modified data source.

The well-known work of Shannon [28] has established an upper bound to the transmission capacity of a noisy channel and the approach to this bound has been the ambition of channel coding and modulation researchers for many years. This continues to the present. Naturally, the characteristics of the mobile radio channel do not permit any close approximation to the Shannon bound; Doppler shifting of received signals due to receiver motion and multi-path propagation, as referred to above, limits radio channel capacity in terms of bit throughput rate (BTR).

Given these constraints it is clear that different services require optimization of transmission parameters to achieve outcomes appropriate to each particular service. As in most engineering situations the optimization process results in compromise. Error free transmission can be achieved by using the automatic repeat request (ARQ) protocol but this creates transmission delays that are acceptable only in particular circumstances. In speech transmissions, the coding and modulation processes deliver output from the transmitter which carries an overhead in terms of the bits which a receiver uses to recover the messages. The result is that while many bit errors are removed by the channel coding some residual errors remain. Nevertheless the recovery of speech frames can still provide acceptable speech intelligibility.

From the standpoint of information theory, as pointed out by Shannon [28], error free transmission is possible provided the entropy of the source is less than the capacity of the channel. The consequence of this constraint is that long data blocks would require very large channel coding support with equally large delays. However in mobile communications the data blocks are relatively short so the transmission strategies outlined above in general meet users' expectations for throughput and data integrity.

There is an extensive literature on all aspects of transmission optimization for different services in mobile communications. Typical examples are available in [29]. Progress from full-rate to half-rate speech codecs in GSM are presented in [30].

As emphasized above, other services require different transmission optimizations. For low-rate video transmission the audio part of the standard is defined by the moving picture experts group (MPEG II) and is also covered in [30]. Low-rate video on mobile radio channels is covered in [31]. To summarize:

3.5.2 Delay sensitive applications

Services which are delay sensitive such as bi-directional speech and video telephony must adopt strategies appropriate to recovering the wanted information from a received transmission which contains residual errors since error-free delivery cannot be guaranteed. A further factor must be taken into account. Mobiles experience deep fades in the received signal that not only result in loss of information bits but also interrupt the synchronization process essential to all framed transmissions. Therefore important information such as the channel equalization bursts in GSM may be lost and the system must be designed to accommodate these eventualities by frequent re-synchronisation which, in turn, sets limits to the coding overhead that can be applied in each transmitted frame. Additionally, speech coding processes need to avoid highly variable bit-rates from the speech coder since it is impossible to predict which bits will be lost in fades.

3.5.3 Bit throughput rate (BTR) sensitive applications

Bit throughput rate is an important factor in delivering services that involve audio and video material in real time. To maintain acceptable quality of service requires a minimum BTR. However, many services do not require bi-directional connections at the same BTR. A user wishing to download material can obtain satisfactory up-link connection at low bit-rate while reserving high bit-rate connectivity for the down-link. In general network operators will aim to maintain a high BTR by supporting utilization of advances in predictive coding techniques and other strategies that support information BTRs. The success of these approaches is readily appreciated by the success of iPhone and iPad services which deliver real-time video material with high quality presentation.

3.5.4 Bit error rate (BER) sensitive applications

In contrast to the foregoing, where real-time exchanges on the radio channel are the requirement, in the case of sensitive data, error-free transmission is essential. Increasingly, users expect to be able to download highly compressed sources, particularly if this involves download of executable software. In these circumstances delay and BTR are less significant provided that the transmitted data are error free. Inevitably, as transmission quality varies in the radio channel, the penalty for error free delivery is variable delay since delay and BTR are inextricably linked.

3.5.5 Layered architecture

It is appropriate in the context of the foregoing to note that increasingly consideration is given to the relationship between the various layers of the open systems interconnection (OSI) model.

Increasingly, communication systems in general and mobile systems in particular draw on a heterogeneous combination of technologies including optical fibre transmission, short range systems such as Bluetooth, Zigbee, and WiFi and mid-range systems such as WiMax. In the classical layered system approach each layer interfaces with the next layer above, but in the heterogeneous environment cross-layer co-operation continues to attract attention. In the context of the present study, however, management of connections and support for mobile communications traffic has relatively little immediate impact on user experience of call quality.

3.5.6 Broadband wireless technologies

The evolution of the various generations of mobile communication systems from 1G through to 4G has been characterized by the achievement of higher and higher bit-rates in the transmission medium. In response to this, services and applications have continued to create demands for yet higher bit-rate capability. Typical systems targeting business users and/or private individuals use a combination of different technologies to satisfy user requirements. A “backbone” optical fibre network brings services to the vicinity of the customer’s premises, while in some cases “fibre to the home” has attracted support since the bandwidth in an optical fibre connection will easily accommodate high-speed internet, telephony and television services. Generally, however, the last segment of the system results from interfacing the optical network with cable facilities typically carrying digital subscriber line (DSL) traffic into buildings. Cable connections can deliver from 150kbps up to 2Mbps on a DSL connection to fixed terminals.

3.5.7 Optical fibre capability

Two considerations influence the completion of service delivery. In remote areas, where optical fibre connections are uneconomic, recourse to satellite services provides a solution. The second consideration relates to the increasing expectation of users for wireless connections to portable user terminals. To meet this requirement several established technologies are available: Bluetooth, WiFi and WiMax.

3.5.8 WiFi

The WiFi acronym derives from “wireless fidelity”. WiFi has become a widely implemented technology for local area networking (LAN) applications. The standard, defined by the Institute of Electrical and Electronics Engineers (IEEE), is accessible under the identifier 802.11 with various versions which have basically similar performance specifications while achieving these using differing air-interface standards.. The standard was formulated to provide high data rate access to services of which internet connection is an important element. Under favourable conditions, WiFi will deliver a data rate of 54Mbps, which exceeds the capability of 3G thanks to using a 20MHz bandwidth, four times the UMTS provision. WiFi greatly exceeds the Bluetooth capability of 2.1Mbps as detailed in the standard 802.15.1. Since WiFi technology has been built into many user devices it represents a most effective means for low cost access to many online services. A typical realization of a WiFi "hot-spot" is shown if Figure 16 [32].

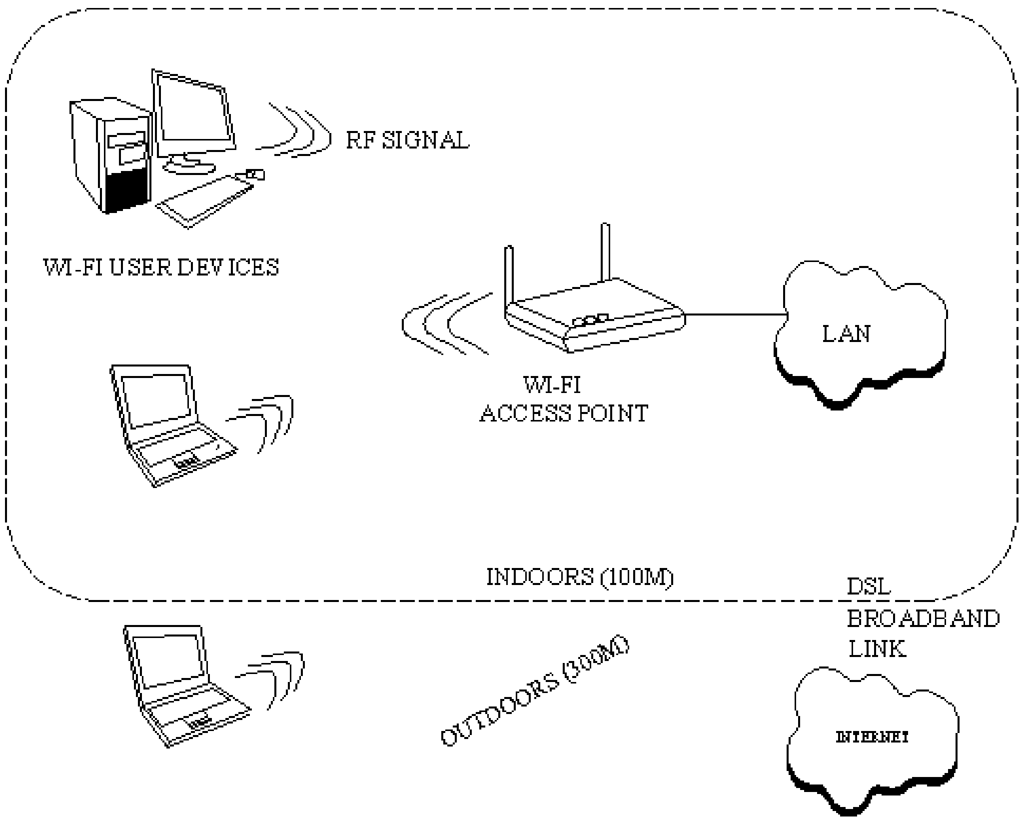


Figure 16: WiFi architecture

Clearly, the limitations of WiFi are two-fold. Range is extremely limited and terminals accessing WiFi must be quasi-static in relation to the base station providing the service. Moving terminals experience some degradation of performance, but movement is limited since WiMax does not support hand-over.

3.5.9 WiMax

3.5.9.1 Origins

The success of WiFi prompted investigation of a solution to the first WiFi limitation of short range. It was recognized that fixed point to point connections could extend the capability of WiFi by enabling WiMax access points to deliver services to many more WiFi hot-spots. Given this objective, opportunities were sought for spectrum availability and the 10-66GHz window was identified as a resource capable of supporting a new service. Accordingly, a standard IEEE 8.2.16 was agreed and developed. In Table 2, the basic parameters of WiMax are compared to those of two of the WiFi standards, 802.11b and 802.11a/g.

Feature	WiMax (802.16a)	WiFi (802.11b)	WiFi (802.11a/g)
Primary application	Broadband wireless Access	Wireless LAN	Wireless LAN
Frequency band	Licensed/unlicensed 2 G to 11GHz	2.4 GHz ISM	2.4 GHz ISM (g) 5GHz U-NII (a)
Channel bandwidth	Adjustable 1.25 M to 20MHz	25MHz	20MHz
Half/full duplex	Full	Half	Half
Radio technology	OFDM (256-channels)	Direct sequence Spread spectrum	OFDM (64-channels)
Bandwidth efficiency	<=5 bps/Hz	<=0.44 bps/Hz	<=2.7 bps/Hz
Modulation	BPSK, QPSK, 16-, 64-, 256-QAM	QPSK	BPSK, QPSK, 16-, 64-QAM
FEC	Convolutional code Reed-Solomon	None	Convolutional code
Encryption	Mandatory- 3DES Optional- AES	Optional- RC4 (AES in 802.11i)	Optional- RC4 (AES in 802.11i)
Mobility	Mobile WiMax (802.16e)	In development	In development
Mesh	Yes	Vendor proprietary	Vendor proprietary
Access protocol	Request/grant	CSMA/CA	CSMA/CA

Table 2: Comparison of WiFi and WiMax

3.5.9.2 The WiMax air interface

An important feature of the WiMax air interface is the implementation of orthogonal frequency division multiple access (OFDMA). This approach relies on division of the available spectrum into a multiplex of sub-carriers each of which carries modulation at a reduced bit-rate. In WiMax, between 128 and 2048 sub-carriers are deployed so that a modest modulation on each sub-carrier results in an overall high bit-rate. The relationship between carrier spacing and modulation spectrum is the basis for the orthogonal relationship among the sub-carriers. The requirement for this is that the sub-carrier spacing should be 1/ the symbol rate. Under this constraint the modulation envelope on each sub-carrier results in a zero component at all other sub-carrier frequencies.

WiMax operating in 20MHz bandwidth in common with WiFi achieves operating bit-rates in excess of 70Mbps; approaching 100Mbps in some cases. However, this capability is not required in many situations, so 1.5MHz channel bandwidth is sufficient in many applications. WiMax therefore supports operation in sub-sets of the sub-carriers providing 128, 512, 1024 or 2048 sub-carriers with equal frequency spacing. The full bandwidth is needed, however, if the WiMax base is supporting a large number of connected users. Additionally, by optimizing the OFDMA capability, a spectral efficiency of 5bps/Hz is achievable in principle, under ideal conditions, although 3.7bps/Hz is more realistic. This compares with 2.7bps/Hz in WiFi a/g.

The higher frequencies above 11GHz present problems for non-line-of-sight applications while the higher end of the 66GHz spectrum can be used for fixed line-of-sight links. Most WiMax applications share the 2-11GHz spectrum with WiFi.

Additionally, the WiMax standard permits implementation of “smart” antennas to obtain beam-forming capability and this results in a greatly extended range of operation, particularly in support of WiFi hot-spots. Generally ranges of 50km are achievable.

With appropriate organization of air interface parameters the range limitations of WiFi are overcome, but there remains the question of delivering services to moving mobiles. This is not practicable in WiFi applications and in the WiMax support for WiFi hot-spots all the participating terminals are static (or “nomadic” when terminals are transported from one fixed location to another).

The question arises as to whether WiMax could support fully mobile terminals. Referring to the sub-carrier spacing it is evident that a disturbance of the sub-carrier frequency raster would impact their orthogonality. This is indeed the case, but the WiMax air interface is robust and some impairment of the sub-carriers fixed spectral location can be tolerated, albeit with some degradation of throughput performance. Expectations are for service delivery at 15Mbps in a 3km radius cell. WiMax with multiple input and output (MIMO) implementation has the added capability to take advantage of the diversity feature of MIMO and utilising maximum ratio combining.

This has resulted in some commentators claiming that WiMax plus 3G is effectively a 4G system. Certainly, OFDM is a candidate for implementation in a future full cellular system but as yet progress is generally referred to as long term evolution (LTE) and the final specification of a next generation system remains to be determined.

3.6 Ultra wideband, an emerging technology

Ultra-wideband (UWB) is a short-range wireless technology capable of transferring wireless data at rates in excess of 100Mbps. One approach is unlike other narrowband wireless technologies; UWB does not modulate the signal with a carrier frequency but instead transmits impulses with durations of less than one nanosecond. The UWB signal is dense in time domain and largely spread in frequency domain. Orthogonal frequency-division multiplexing (OFDM) offers an alternative to achieving ultra-wide transmission bandwidths.

In February 2002, the Federal Communications Commission (FCC) authorised the unlicensed use of UWB in 3.1-to-10.6GHz with a power cap of -41dB [33]. The UWB signal should not interfere with other wireless standards as the signal power is spread over a large bandwidth. The main application of UWB technology is in short-range wireless networking. UWB can replace the data cables in the home or business that are used for transferring digital data at high data rates. The low power operation of the UWB transceivers promises a dominant position for UWB systems in home networking. Intel has already started an intensive research programme to develop UWB transceivers to be supplied with Intel’s next generation chip-sets [34].

In addition to its high speed and low power operation, UWB offers simpler transceiver architecture than narrowband wireless systems. Single channel UWB transceivers do not incorporate any mixer and oscillator as no frequency down conversion is required. The simplicity of the UWB architecture results in a very low cost for design and fabrication of UWB prototypes. However, orthogonal frequency-division multiplexing (OFDM) requires more sophisticated transceiver architecture [34].

The specification of required broadband amplifier can be extracted from the UWB technology characteristics. The bandwidth of the amplifier needs to be 3-10GHz to completely recover the UWB signal. Typically a power gain larger than 10dB is necessary as a very weak UWB signal is received by antenna. As the amplifier is the first stage of the system, the broadband amplifier should not add significantly to the signal's noise.

3.7 Mobile channels

The performance of any communication system is ultimately determined by the medium used. This medium, whether it be an optical fibre, the hard-disk drive of a computer or a wireless link, is the communication channel. A wide variety of channels exists, which may be divided into two groups. A wired channel is when there is a solid connection between transmitter and receiver. A wireless channel is when there is not.

Wireless channels tend to be more unreliable than wired channels. In wireless channels the state of the channel may change within a short period of time, rendering communication difficult. Wireless channels may be further distinguished by their propagation environment – such as urban, suburban, indoor, underwater or orbital [35].

This research focuses on the factors that influence the performance of wireless channels. It considers analytic models of basic propagation effects encountered in wireless channels and shows how they translate into the performance of different communication systems. This knowledge is crucial in order to design and parameterize simulation models of wireless channels. A different area where this knowledge is important is the design of communication protocols. In general, the research addresses the reader who is interested in the analysis behind the wireless channel.

It is not always clear what is referred to as a wireless channel in a communication system, since there are multiple processes in the transmission and reception of a signal. Figure 17 [36] illustrates the propagation, radio, modulation and digital channels.

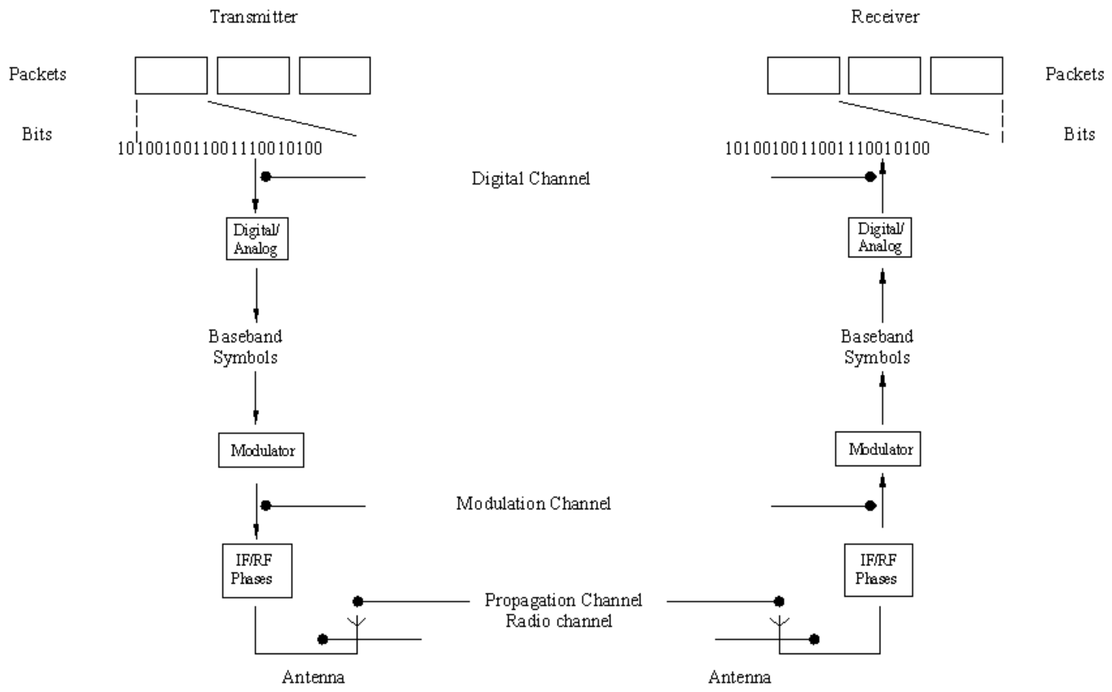


Figure 17: Propagation, radio, modulation and digital channels

3.7.1 The propagation channel

The propagation channel, which lies between the transmitter and receiver antennas, is influenced only by the phenomena that influence the propagation of electromagnetic waves. It is almost always linear and reciprocal. All aspects of this channel affect only the attenuation of the transmitted signal. This channel therefore has a multiplicative effect on the signal. The signal transmitted consists of the information modulated on top of the carrier frequency [35].

3.7.2 The radio channel

The radio channel consists of the propagation channel and both the transmitter and receiver antennas. As long as the antennas are considered to be linear, bilateral and passive, the channel is also linear and reciprocal. The signal is affected only by attenuation; however the attenuation of the propagation channel might be different since it might be modified by the used antennas, where the antenna influence is strictly linear [35].

3.7.3 The modulation channel

The modulation channel consists of the radio channel plus all system components (such as amplifiers and different stages of radio frequency circuits) up to the output of the modulator on the transmitter side and the input of the demodulator, including the decision process, on the receiver side. Whether the system is linear depends on the transfer characteristics of the components between demodulator or modulator and the antennas. The channel is also non-reciprocal, because amplifiers (the system component added to the radio channel) are non-reciprocal. The process of amplifying the received signal at this point means that noise and interference may damage the signal. They may already be present at the radio channel; however, especially noise from electric circuits is added at this channel level, so a complete characterization of the additive effects cannot be done at the radio channel level. The signal consists here of the baseband symbols, which are modulating a carrier frequency [35].

3.7.4 The digital channel

The digital channel consists of the modulation channel plus the modulator and demodulator. It relates the digital baseband signal at the transmitter to the digital signal at the receiver, and describes the bit error patterns. The channel is non-linear and non-reciprocal. At this channel level no further effects come into play; instead the corrupted signal is interpreted at this level as bit sequence. If the signal has been corrupted too heavily, the interpreted bit sequence differs from the true bit sequence it is intended to convey.

The inputs to this channel are bits, which might stem from packets. The bits are grouped and then turned into analogue representations, so called symbols. These belong to the baseband. This analogue signal is then passed to a modulator, which modulates these baseband signals on to the carrier frequency.

Assuming ideal antennas, the propagation channel becomes identical to the radio channel. The radio channel attenuates the received signal by a time-varying factor. This attenuation might be compensated by the modulation channel, since amplifiers are employed here to boost the received signal. However, at the modulation channel, random, time varying noise also enters the system. This adds a distorting element. If the attenuated signal is largely amplified, the noise will also be amplified strongly. Therefore it is up to reliable detection methods in the digital channel to extract the transmitted signal from the noise.

In addition to noise, which is always present at this stage of a communication system, electromagnetic waves from other communicating devices may interfere with the received signal. This interference significantly affects the performance, like noise. The interfering signal is also time variant.

Two performance metrics – the symbol error probability (SEP) and the bit error probability (BEP) – are commonly used with a digital communication system. Both relate to the digital channel; the BEP to the interpreted bit stream and the SEP to the stream of symbols still to be interpreted. Both depend on the instantaneous power ratio between the received signal power and the noise and interference powers. This instantaneous power ratio is given by the signal-to-noise-and-interference ratio (SNIR).

A varying SNIR can sometimes result in a varying SEP or BEP. If the average SNIR of a link is available, the average error rates like symbol error rate (SER) or bit error rate (BER) can be obtained. In general, the relationship between SNIR and error rates or error probabilities depends on many factors. A profound understanding of the influence of all the effects regarding the receiver SNIR is therefore needed to investigate the performance of any wireless communication system [35].

3.8 Path loss, shadowing and fading

Three effects result in an overall attenuation of the transmitted signal: path loss, shadowing and fading.

Path loss depends on the distance between the transmitter and the receiver. It plays an important role in larger time scales like seconds or minutes, since the distance between transmitter and receiver in most situations does not change significantly in smaller time scales.

Shadowing varies on the same time scale as path loss and causes fluctuations of the received signal strength at points with the same distance to the transmitter. However, the mean over all these points yields the signal strength given by path loss only.

Fading leads to significant attenuation changes within smaller time scales such as milliseconds or even microseconds. Fading is always caused by a multi-path propagation environment – therefore by an environment reflecting the transmitted electromagnetic waves such that multiple copies of this wave interfere at the receiving antenna [35].

CHAPTER 4

Voice quality

4.1 The significance of voice quality

Voice quality is a significant feature of speech. It is used to convey various aspects of conversations such as emotions, intentions and the mood of the speaker. In wireless networks, it is a major index that determines the overall appeal of a telecom network to subscribers and a major determinant of their confidence in the particular operator providing the voice services.

Customer satisfaction is key to the network operator's commercial success. Therefore, it is crucial to measure and to evaluate the quality of speech in live networks either through commonly applied drive tests or other kinds of field trials. Thus, providing clear, uninterrupted voice is a critical factor in the successful operation of telecommunications networks. To maintain customer satisfaction, mobile operators are dedicating drive test tools such as TEMS (a technology used by wireless operators to measure, analyse and optimise their mobile networks) to measure different network parameters as perceived by users, such as single strength, drop call rate, successful call rate, hand-over call rate and voice quality. TEMS offers the speech quality index (SQI) quality measure for estimating the down link speech quality in GSM, WCDMA or CDMA cellular networks. [37]

Because providing clear, uninterrupted voice is a critical factor in the successful operation of telecommunications networks, it is important for the operator to maintain control of network voice quality. Voice quality measurement techniques and algorithms have a long history. Although much progress has been made in the development of objective testing of voice quality, much work is needed to obtain a comprehensive means of describing how subscribers experience the quality of the mobile network.

The importance of voice quality affects every aspect of emerging telecommunications technology. Depending on the kind of technology, different levels of complexity and sophistication are required to sustain and improve the quality of voice connections. To this end, understanding different voice quality levels in every technology is essential to enable network operators to evaluate quality and means of improvement.

In the context of this thesis, the requirement of assessing voice quality is particularly demanding. As argued later, operators need to take account of call voice quality in order to adjust their call charging regimes according to the quality of the service they deliver. Ideally, this would require that the quality of the network connection be monitored and the call charges adjusted on a frame by frame basis. Given that this is impractical with current technology, alternatives must be found. It is widely appreciated that bit error rates are related to call quality in that high rates of bit errors do correlate with poor voice quality. However, with increasingly sophisticated signal processing, bit errors in a wireless connection can be disguised to a certain extent.

It is appropriate to review speech quality measures to examine the extent to which each might be applicable in the way in which call charging can be determined by call quality.

4.2 Understanding call quality

Call quality testing has traditionally involved picking up a telephone and listening to the quality of the voice. However, requiring listeners to evaluate calls over many repetitions is subjective and can be difficult and expensive to set up and execute. Considerable progress has been made in establishing objective measurements of call quality. Among the standards developed are:

- PSQM (ITU P. 861, 02/98) / PSQM+ - perceptual speech quality measure;
- MNB (ITU P. 861, 02/98) - measuring normalizing blocks;
- PESQ (ITU P. 862, 02/01) - perceptual evaluation of speech quality;
- PAMS (British Telecom) - perceptual analysis measurement system [38].

PSQM, PSQM+, MNB and PESQ are part of a succession of algorithm modifications starting in ITU (International Telecommunication Union) standard P. 861, of February 1998, dealing with objective quality measurement of telephone band 300Hz-3400Hz speech codes. As described later, P. 861 was withdrawn to be replaced, on 23 February 2001, by P. 862, which deals with the perceptual evaluation of speech quality (PESQ) – an objective method for end-to-end speech quality assessment of narrow band telephone networks and speech codecs [39]. British Telecom developed PAMS, which is similar to PSQM. The PSQM and PAMS measurements send a reference signal through the telephone network and then compare, using digital signal processing algorithms, the reference signal with the signal received on the other end of the network.

However, these approaches are not really suited to assessing call quality on a data network. They cannot fully take into account such issues as delay, jitter and datagram loss.

4.3 Overview of speech quality measurement

Speech quality can be measured by subjective testing or by objective testing.

4.3.1 Subjective testing – mean opinion score

Mean opinion score (MOS) provides a numerical indication of the perceived quality of received human speech over the connection. The MOS is expressed as a single number between 1 and 5, where 1 is lowest perceived quality and 5 is the highest. The MOS is generated by averaging the results of a set of standard, subjective tests, where a number of listeners rate the heard audio quality of test sentences read aloud by both male and female speakers. The listener must rate each sentence according to whether it is:

Rating	**Perceived quality**
5	Excellent
4	Good and not annoying
3	Fair, slightly annoying
2	Poor, annoying
1	Bad, very annoying

Table 3: Perceived quality rating of human speech

The MOS is the arithmetical mean of all the individual scores. MOS tests are specified by International Telecommunication Union (ITU-T) recommendation P.800 [40].

4.3.2 Objective testing

The results of MOS testing are expensive and impractical for testing in the field, since the testing takes much time and resources. Automatic testing algorithms were created in an attempt to formulate objective network testing similar to signal to noise ratio (SNR), bit error rate (BER) and received signal strength indictor (RSSI), which are used to measure the signal quality.

Perception models for evaluating speech quality were developed by teams led by Mike Hollier, at BT Laboratories [41], and by John Beerends and Jan Stemerdink, at KPN Research [42], which led to subsequent innovations in the 1990s on the use of perception for voice quality assessment. Hollier observed that taking account not just of the amount of audible distortion, but also its distribution, could make quality predictions much more accurate. His work was taken up in 1996 by Antony Rix and forms the core of the perceptual analysis measurement system (PAMS) [43].

4.3.3 Perceptual speech quality measure (PSQM)

Perceptual speech quality measure (PSQM) is an algorithm originally developed to test voice quality through a voice encoder/decoder (codec). PSQM was introduced by Beerends and Stemerdink and was adopted in 1996 as ITU-T recommendation P. 861.

PSQM is an intrusive quality measurement system that involves the comparison of a voice sample, after it has passed through the codec, with the original sample. This standard is currently being used to test voice quality over end-to-end voice paths, even though the standard cannot account for errors associated with packet delay, packet jitter, lost packets, duplicate packets, noise or other factors that might affect voice quality in a VoIP (voice over internet protocol) environment. It therefore became clear that PSQM was not suitable for testing networks, where speech codecs are only one part of a complex chain. PSQM was found to correlate very poorly with subjective opinion in some commonly occurring situations of speech clipping, background noise, packet loss in VoIP networks, filtering in analogue elements (such as handsets or two-wire access loops) and variable delay (common in VoIP). The correlation achieved by PSQM against subjective MOS was only 0.26 whereas an ideal model would have a correlation of 1 [44].

4.3.4 Perceptual analysis measurement system (PAMS)

Mike Hollier developed perceptual analysis measurement system (PAMS) as an extension to the bark spectral distortion model – the first objective measure to incorporate psychoacoustic responses, and based on the assumption that speech quality is directly related to loudness. PAMS was the first model to focus on end-to-end behaviour, including the effect of filtering and variable delay. It is designed to give an automated voice-quality rating, based on a comparison between the received voice signal and the original signal. Designed for network-wide testing, it can take account of delay, jitter and other events that would affect voice quality in a VoIP network.

BT aimed to produce a model suitable for end-to-end network testing. PAMS was therefore designed from the start to include analysis components for level and time alignment. These are missing from PSQM, had to be provided separately and could have a significant impact on the model's performance. To facilitate this development, a large database of subjective tests was assembled. The database contains a wide range of codecs, errors and packet loss, and noise conditions. It is believed to be the largest of its kind and contains more than 25,000 distorted speech recordings and more than 250,000 subjective votes.

Version 1 of PAMS was released in August 1998 and provided better performance than PSQM in conditions with noise, codecs or packet loss. It was extended to take account of variable delay in version 2 which, released in December 1998, was the world's first model suitable for assessing VoIP. Version 3, released a year later, was the first model able to assess the full range of conditions, including VoIP and analogue networks [44].

4.3.5 Perceptual evaluation of speech quality (PESQ)

Perceptual evaluation of speech quality (PESQ) is the new ITU-T standard (ITU-T P. 862) for measuring the voice quality of communication networks. It is expected eventually to replace PSQM. It builds on the PSQM and PAMS algorithms by adding additional processing steps to account for signal-level differences and the identification of errors associated with packet loss.

In parallel with the development of PAMS, BT and a number of other organizations pressed the ITU-T to select a replacement for PSQM that would be more suitable for testing networks. In order to achieve this, ITU-T study group 12 staged a competition between September 1998 and March 2000. BT (with PAMS), KPN (with PSQM99, an extended and improved version of PSQM), Ascom, Ericsson and Deutsche Telekom took part.

The outcome was a division of the models into two groups. The winners were PAMS and PSQM99 but there was statistically no single winner. PSQM performed better on certain conditions of rapid gain variations and severe temporal clipping whereas PAMS performed better on conditions of VoIP and filtering. The second group all had significantly lower average correlation and showed shortcomings on many more of the condition types. PSQM had poorer performance still. It was therefore decided to integrate the best two models, PAMS and PSQM99, into a single model. The ITU-T decided that this model, to be accepted, would have to outperform both PAMS and PESQ. The BT group collaborated with KPN to achieve this. The result was PESQ.

In May 2000 PESQ passed all the new performance criteria and was submitted for standardization as P. 862. This process was completed in February 2001 with the final approval of P. 862 and the withdrawal of P. 861 [44].

4.4 Recognizing GSM digital speech

Digalakis, Neumeyer and Perakakis [45] distinguished three possible answer seizure ratio (ASR) architectures, depending on how the speech recognition processing is distributed between the mobile terminal and the machine running the ASR application, i.e., between the client and server sides. The three are: local recognition (client-only processing); remote recognition (server-only processing); and distributed recognition (client-server processing).

Local recognition. The best way to avoid both coding distortion and transmission errors is to perform the speech recognition at the user local terminal. In this case, speech coding distortion and transmission errors are not a problem. Nevertheless, this approach has two important disadvantages: first, the application must reside at the local terminal, which must support the whole computational load; and secondly, it is not possible to reproduce the speech signal at the remote end.

Remote recognition. The best alternative to reduce the computational load at the local terminals is to let the server perform all the recognition process. In this case, voice should be transmitted over the mobile network and consequently will be affected by the already mentioned distortions. In this case, the mobile terminal has to know nothing about the kind of application running at the remote end, the only requirements being the use of a standard codec supported at both ends. This fact becomes relevant for the design of applications that integrate voice, data or any other kind of media, since it allows universal access from almost any terminal.

Distributed recognition. A compromise between local and remote recognition consists of performing part of the recognition process at the client end (namely, the parameterization) and the remaining part at the server end (see for example, [30] or [39]). The advantages of this approach are that the bandwidth required to transmit the recognition parameters is small and the computational effort needed for the parameter extraction is not so high.

However, a standardized front-end is needed so that every client terminal computes and transmits the same parameters. Moreover, it is impossible to reproduce or process the speech signal at the remote end. And finally the requirements on the terminals regarding remote recognition are higher.

This alternative seems promising; much effort is being made to address its shortcomings (see [46] for example). Nevertheless, this research explores the remote recognizing alternative, since it seems to offer two relevant advantages: first, it neither imposes restrictive conditions on the client terminal's capabilities nor creates the need for special setting or agreements between client and server; and secondly, it preserves the transmission bandwidth requirements and the compatibility with the existing standard-based voice applications [47].

4.5 Remote recognition: key issues

As previously stated, the GSM environment entails three main problems for ASR systems: noisy scenarios, source coding distortion and transmission errors. Other sub-systems of the GSM system – for example, the discontinuous transmission (DTX) or the insertion of comfort noise – can also affect the performance of a remote ASR system. In intervals between speech bursts the channel becomes slient giving the impression that the call has been lost. Adding a small amount of background noise overcomes this. Discontinuous transmission, although conservatively designed, occasionally clips the speech signal. The insertion of comfort noise slightly disturbs the estimation, at the remote server, of the background noise characteristics. Other parts of the system having to do with the signaling process, such as the hand-over protocol, can also sporadically affect the performance of an ASR system [47].

The following three subsections present the three main above mentioned problems: noisy scenarios, source coding distortion and transmission errors. There is a particular focus the last two, as they have previously received much less attention and constitute the main motivation of this work.

4.5.1 Noisy scenarios

GSM allows the user to make calls from almost anywhere. Reliable ASR systems should therefore be robust to any kind of background noise as well as to the Lombard effect – the involuntary tendency of speakers to increase their vocal effort when speaking in loud noise to enhance the audibility of their voice.

Current research addressing the noise problem is focusing on speech enhancement, more robust parameterization techniques and techniques to adapt clean-speech hidden Markov models (mathematical formalisms that allow modelling of a stochastic system, which may undergo characteristic changes at uncertain times) to noisy speech conditions.

4.5.2 Source coding distortion

At the typical rates of operation of cellular systems, codecs based on the source-filter model are used most of the time. These codecs achieve their medium or low bit rates by assuming a simplified speech production model with negligible interaction between source and filter. The filter is determined on a frame-by-frame basis while the excitation is computed with a higher time resolution (from two to four times per frame, depending on the codec) usually by means of an analysis-by-synthesis procedure aimed at minimizing a perceptually weighted version of the coding error. As a result, these codecs introduce two types of distortion, one caused by the quantization of the parameters to be transmitted and the other being the noise arising from the inadequacy of the model itself. Consequently, the waveform, short-time spectrum, and other relevant characteristics of the encoded and decoded speech signal, are somewhat different from those of the original one.

Since speech codecs for mobile telephony operate, in general, below 16 kilobits per second (the half and full-rate GSM codecs work at 5.6 and 13 kb/s, respectively) the speech coding distortion will significantly affect the recognition performance.

To evaluate a robust parameterization – namely RN LFCC (root-normalized linear frequency cepstral coefficients) in the GSM environment, including the source coding effects – several speech recognition experiments have been performed with speech coded by the full-rate (FR) and half-rate (HR) standard codecs. The experiments revealed that the recognition losses are significant in both cases, but more important in the HR codec.

The effect of tandeming, addressed by Lilly and Paliwal [48], deserves separate comment for two reasons: first, its influence on recognition performance is serious; and secondly, tandeming is very common in practice.

Once the speech signal has been encoded there is no real reason – assuming that the entire communication network is digital – to decode it until it reaches the end-user. But this is not how things work out in practice. When the signal goes through international links, it is usually decoded and re-encoded to undergo the international segment, to be decoded and re-encoded again when it enters the mobile network. Even when the signal does not cross political borders, it occasionally suffers the same tandeming process when the near- and far-end telephone operators are different. For networking reasons, and more frequently than suspected, the speech signal may go through two or more encoding-decoding stages.

Therefore, a realistic evaluation of the influence of speech codecs on remote ASR systems should consider tandemings.

4.5.3 Transmission errors

Transmission errors are an inevitable part of GSM and so should be included in the benchmark experiments.

GSM, like every mobile phone system, provides a mechanism to protect the speech signal against transmission errors – the channel coding. More specifically, the channel encoder (which is explicitly designed for each standard codec) classifies the source bits in several categories depending on their relative perceptual impact, as commented on later. In this way, not only are some errors detected and even corrected, but their influence on the (decoded) speech perceptual quality is also minimized. Nevertheless, the GSM channel coding is not capable of detecting and correcting all the errors, and some may be present in frames labelled correct [47].

4.6 Summary of voice quality measures

Although much progress has been made in developing objective measures of speech quality, there remain obstacles to implementing these during the progress of a conversation over a communication channel which involves the variability of the radio channel. Possibly, by deploying speech recognition techniques, an alternative to real-time speech quality measurement has the potential to enable telecommunications network operators to monitor the speech connections they support and achieve the objective of making call charges directly related to voice quality.

For the present, the emphasis must remain focused on high level functionality in which call monitoring makes sure that call outages due to the rapid and deep fades are recorded and that otherwise bit rates in successful connections are tracked to provide frame by frame a measure of connectivity. This will provide a basis for call charging as a function of call quality.

CHAPTER 5

Symbian architecture and QMeter® parameters

5.1 Introduction

Early mobile phones did not require complex operating systems or software development platforms. Many of today's mobile phones, in contrast, are able to run downloaded applications. They need a more advanced operating system to provide a reliable and versatile platform for third party software.

Symbian Ltd was founded in 1998 to provide a more advanced platform for data enabled smart-phones. A spin-off of Psion plc, it initially included two of the world's largest handset makers, Nokia and Ericsson. Motorola joined later that year and Matsushita and Siemens became shareholders six years after that. By summer 2007, the Symbian Ltd shareholders were Samsung, Ericsson, Sony Ericsson, Panasonic, Nokia and Siemens. Nokia became the sole owner of Symbian in 2008.

Symbian was the most popular smart-phone platform between 2002 and 2010, when some 450 million devices were estimated to be using it. But from 2007 onward it lost market share and developer loyalty with the introduction of the iPhone and Android platforms. Early in 2011, Nokia announced that it was abandoning Symbian [49].

Symbian is a proprietary operating system, designed for mobile devices, with associated libraries, user interface frameworks and reference implementations of common tools. Structured like many desktop operating systems, Symbian's major advantage is that it was built specifically for handheld devices, with limited resources that may be running for months or years [50]. Many system functions have been pushed out to user-space, so that much less memory is taken up on boot and their structure is more dynamic. It was also designed to consume less power. For example, Symbian switches off peripherals when they are not being used [51]. Symbian was developed with reliability, fault tolerance, power consumption and read only memory (ROM) utilization in mind, as well as the integrity and security of the user's data [52].

Architecture is at the heart of satisfying customer demands. If the operating system is unable to provide the technology that customers need, they will build their own extensions or license them from elsewhere. The operating system will then suffer from incompatibilities and missing technologies. Charles Davies calls this problem "defining the skin" – understanding, maintaining and managing the bounds of the system [51].

Crucial to the successful evolution of the operating system are: managing the requirements push from customer products; managing external pressure on the system to evolve; and managing licensees and partners to create their own extensions to the system [51].

5.2 Symbian architecture

The Symbian operating system architecture includes: the kernel services and hardware layer; the base services layer; the system layer; the application services layer; and user interface software and applications.

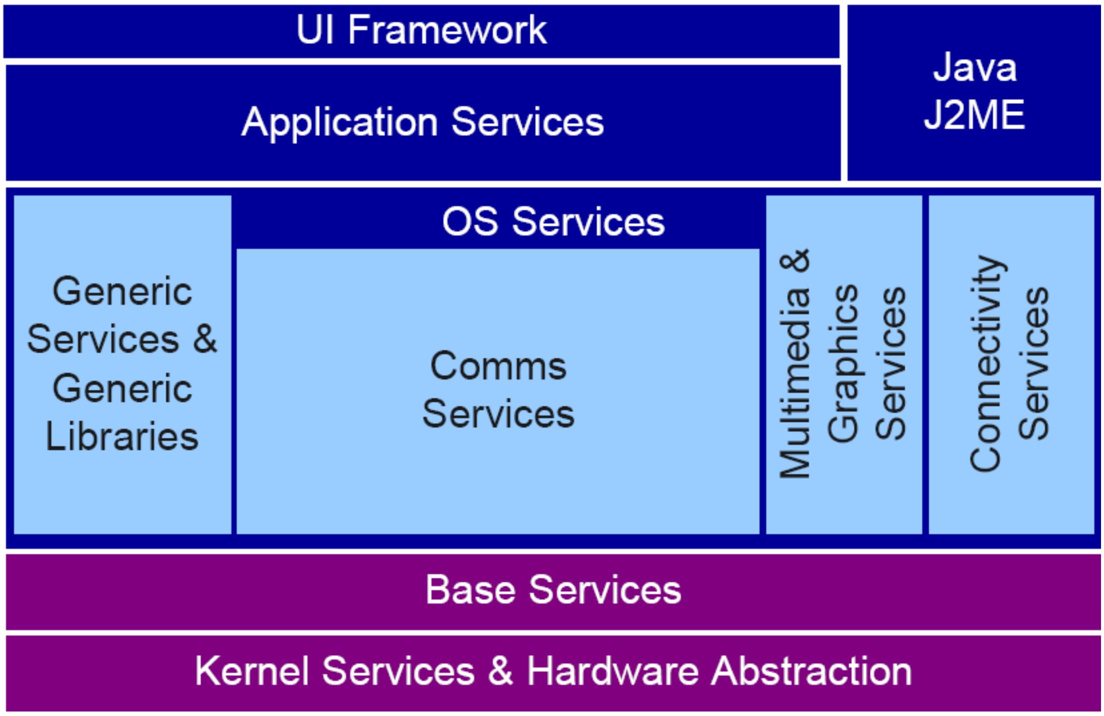


Figure 18: Symbian operating system architecture [51]

5.2.1 Kernel services and hardware interface layer

The kernel services and hardware interface layer is the lowest layer of the Symbian OS. It contains the operating kernel itself, file server, memory management and device drivers. It also embraces the supporting components that abstract the interfaces to the underlying hardware, including logical and physical device drivers and a “variant support”, which implements pre-packaged support for the standard, supported platforms [53].

Before the release of Symbian OS v8, the kernel was the EKA1 (kernel architecture 1) – an original Symbian OS kernel. In Symbian OS v8, the EKA2 (kernel architecture 2) real-time kernel was used as an option for the first time. Symbian OS v8.1a and Symbian OS v8.1b are designated as Symbian OS v8.1 release, with the original kernel architecture. From Symbian OS v9 onwards, the EKA2 kernel has always been used [51].

5.2.2 Base services layer

The base services layer – the foundation layer of the Symbian OS – is the lowest level reachable by user-side operations. It includes: the file server and the user library; the plug-in framework, which manages all Symbian OS framework plug-ins; Store, which provides the persistence model; the central repository; the database management system (DBMS) framework; and the cryptography library. The base services layer also includes the text window server and the text shell, the two basic services from which a completely functional base port can be created without the need for any higher layer services [54].

5.2.3 OS services layer

The system layer – the middle layer of the Symbian OS – includes a set of frameworks and libraries that are not critical for the functioning of a device, but without which the device could not do anything useful without dramatic effort. They extend the bare system into a complete operating system.

The system layer includes libraries and services providing access to telephony, networking, windowing, multi-media and so on. These components execute as processes within the user memory context and are accessed through approved kernel-mediated interprocess communication (IPC) mechanisms.

The services at this layer are divided into four major blocks: generic operating system services; communication services; multi-media and graphics services; and connectivity services. These are technology specific but application independent services in the operating system.

The following servers are found in this layer:

- *communications framework* – the communications root server and ESock (sockets) server, which provide the foundation for all communication services;
- *telephony* – the ETel (telephony) server, fax server and other principal servers for all telephony-based services;
- *networking* – the TCP/IPv4/v6 networking support;
- *serial communications* – the C32 (serial) server, standard serial communications support;
- *graphics and event handling* – the window server, font and bitmap server, which provide all screen drawing, font support, system and application-event handling support;
- *connectivity* – the software install server, remote file server and secure back-up socket server, which provide connectivity services support; and
- *generic* – the task scheduler, which provides scheduled task launching.

The other important frameworks and libraries found in this layer are: the multi-media framework, which provides support for cameras for moving-image recording, replay and manipulation, and audio players; and C-standard library, a support library for software porting [55].

5.2.4 Application services layer

The application services layer sits above the system layer. Providing developers with a platform to develop user interfaces to the data, the application engines can either be from the manufacturers or from third party developers.

The application services layer provides support independently of the user interface for applications on the Symbian OS. These services are categorized into three broad groups:

- *System-level services*. These are used by all applications; for example, application architecture or text handling.
- *Services that support generic applications and application-like services*. These include, for example, personal productivity applications (vCard and vCal, Alarm Server) and data synchronization services (OMA Data Sync). They cover a number of key application engines which are used and extended by licensees (the calendar and agenda model) as well as legacy engines such as the data engine, which licensees may choose to retain.
- *Services that support generic but application-centric technologies*. These include, for example, mail, messaging and browsing (messaging store, multipurpose internet mail extensions recognition framework, hypertext transfer protocol transport framework).

Applications in the Symbian operating system broadly follow the classic object-orientated model-viewer-controller pattern. The framework level support encapsulates the essential relationships between the main application classes and abstracts of all the necessary underlying system level behaviour. In principle, a complete application can be written without any further direct dependencies.

The application services layer shows how the OS as a whole has evolved. It contains application engines needed by all the devices. It also contains a small number of application engines that are mostly now out of date. These include, for example, the what-you-see-is-what-you-get (WYSIWYG) printing services and the office application engines, including sheet engine, a full spreadsheet engine more appropriate for personal digital assistant (PDA)-style devices. Symbian OS v9.3 provides the session initiation protocol (SIP) framework – the foundation for the next generation of mobile applications and services [56].

5.2.5 User interface layer

The user interface layer – the top layer of the Symbian OS – consists of various user applications and frameworks. They include both those installed by the original equipment manufacturer – such as the Series 60 platform, UIQ and Foma – and after-market applications installed by the end user.

Applications at this layer access device-specific functionality exclusively through OS-provided services. As such, applications written against a specific software development kit can be compiled for, and run on, a wide variety of supported phones. The end user generally associates these components with the device because they perform useful and visible tasks.

The user interface layer architecture in the Symbian OS is based on a core framework, named Uikon, and a class hierarchy for user interface controls, named the control environment. Uikon was created to support easier user interface customization, including “pluggable” look-and-feel modules [57].

5.3 QMeter® parameters

QMeter® is the set of Symbian applications named signal meter (SM), for measuring call quality, and bandwidth meter (BM), for bandwidth quality measurement.

5.3.1 Signal meter parameters

The QMeter® is developed in Symbian C++ on the s60 third edition platform. The call quality parameters were computed on a bundle of 10 calls. Signal strength was captured for every five milliseconds during the active call and the average signal strength was calculated at the end of the call [58]. The signal strength quality is categorized based on the scale below for the average signal strength of an individual call.

Signal strength range	**Quality categorization**	**Score**
-120.00dB to -95.00dB	Extremely bad	1
-95.00dB to -85.00dB	Bad	2
-85.00dB to -75.00dB	Average	3
-75.00dB to -65.00dB	Good	4
- 65.00dB to -55.00dB	Very good	5

Table 4: Signal strength, quality categorization and score

5.3.2 Signal strength measurement

The QMeter® listens for outgoing and incoming calls. The following quality calculations are measured and logged:

- the number of successful and unsuccessful call attempts made for every 10 call attempts;

- the successful and unsuccessful call attempts, classified on whether the call is successfully connected by the network;
- the call drop information, such as calls normally dropped from either party or dropped because of hand-over during the cell change;
- the number of normal dropped and hand-over dropped calls, with their average scores.

The successful call attempts rate and successful hand-over rate and their scores are calculated on a scale of 1 to 5. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The successful call rate score is computed based on the number of calls that are successfully connected by the network in a bundle of 10 calls. Similarly, the unsuccessful call rate score is computed based on the number of unsuccessful calls in a bundle of 10. The successful call rate is calculated based on the scale below.

Score	**Successful call attempts**
1	1-2
2	3 – 4
3	5- 6
4	7 – 8
5	>8

Table 5: Score versus successful call attempts

Normally dropped calls are categorized based on the call drop information. If the call is dropped from either party, it is categorized as normally dropped. At the end of the tenth call in the bundle of 10, the normally dropped rate score is computed based on the number of calls normally dropped. The average signal strength of the normally dropped calls is computed by averaging the signal strength of the individual normally dropped calls. The normally dropped call rate is scored as below:

Score	**Normally dropped calls**
1	< 4
2	< 4 & <6
3	< 6 & <7
4	< 7 & <8
5	>8

Table 6: Score versus normally dropped calls

Hand-over dropped calls are categorized based on calls dropped because of hand-over during the cell change. At the end of the tenth call in the bundle of 10, the hand-over dropped calls rate is computed. The average signal strength for the hand-over dropped calls is computed by averaging the signal strength of individual hand-over dropped calls. The call quality is derived from adding together the average signal strength score of all successful calls, the successful call rate score and the hand-over success rate score. The table below depicts the final call quality for a bundle of 10 calls:

Score	**Classification**
<1	Extremely bad
1 – 2	Bad
2- 3	Average
3- 4	Good
4 – 5	Excellent

Table 7: Call quality scores

5.3.3 Bandwidth meter parameters

Bandwidth parameters affect the quality of download as experienced by the end-user. To calculate bandwidth:

1. When a new download is initiated (counters start to update in 10 seconds) save the current time as T_b.
2. When the download is finished (counter is not updated in 10 seconds) save the last valid counter time as T_e and the total bytes received between T_b and T_e as B_r.

3. Calculate the average speed for this download as $S_m = B_r/(T_e - T_b)$.

4. If average speed is calculated for 10 downloads: $S = (S_1 + S_2 + ... + S_{10}) / 10$.

Here is the method to calculate the average internet download speed:

- Average speed > 256kbps: 5 (Excellent)

- Average speed >= 128 & <256kbps: 4 (Good)

- Average speed >= 64 & <128kbps: 3 (Average)

- Average speed >= 32 & <64kbps: 2 (Bad)

- Average speed < 32kbps: 1 (Very bad)

CHAPTER 6

QMeter® algorithm

6.1 Introduction

This chapter discusses QMeter® algorithms, signal meter, bandwidth meter and their modules with flow charts. The QMeter architect is Akram Aburas, who conceptualized and designed the QMeter system. The development on Symbian C++ was done by Webgate, a Bulgarian company.

6.2 Signal measure algorithm

The signal measure programme is developed in developer platform s60 third edition of Symbian. It is sophisticated and able to accept the user preference of selecting the input. The programme accepts two kinds of settings – log change and log location. In log change, the user can select to log either for every five milliseconds or when signal strength is changed. The idea behind using these two options is to address the problem of the increasing size of the log file when the signal strength is being logged for every five milliseconds. In log location, the user can select either internal memory or memory card.

The system works during the active call, which also records the GPS co-ordinates during the call. The system provides for an auto-start, which helps the user to log and locate the information of all the active calls on the map, once the auto-start is enabled. This reduces the problem of starting the application every time during the active call. It also helps to analyze the signal quality at any location for a given operator.

It supports both internal and external GPS connected through Bluetooth. The captured GPS co-ordinates are used for plotting the average signal strength on the map.

The system also records the number of successful and unsuccessful calls made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. Call drop information, such as calls dropped from either party or because of hand-over during the cell change, is also recorded. The number of normal dropped and hand-over dropped calls, with their average scores, is also recorded. The landmarks marked in red are calls dropped due to hand-over; the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls [59].

A generic algorithm of signal measure is presented below. Note that the loop from line 5 (While) to line 13 (End of while) would execute only once, and hence increment counters only once, because of the single call. But the loop would execute multiple times until the call is active and logs the signal strength information.

1. Get the preferences for log_change, log_location
2. Get total_calls, call_attempts_failed, call_attempts_successful, normal_dropped_calls, hand-over_dropped calls
3. If (total_calls=10) reset all variables to zero
4. If (call_attempt=failed)

   total_calls=total_calls+1

   call_attempts_failed=call_attempts_failed+1
5. While (phone_status != idle && call_attempt = successful)
6. Total_calls=total_calls+1
7. Call_attempts_successful=call_attempts_successful+1
8. If(gps_coords available)

Get the gps_coords

9. Get the date, time, cell_id.
10. Get the signal_strength
11. If (log_change=5ms)

    Write (“signal_measure.log”, date, time, gps_coords, cell_id, signal_strength)

12. If (log_change=when changed) and (signalstrengthchange=yes)

    write(“signal_measure.log”, date, time, phone status, signal strength)

13. End of while
14. End of call
15. Calculate average_signal_strength
16. If (average_signal_strength <= -95 && average_signal_strength >= -120) Signal quality=Extremely bad

    Elseif (average_signal_strength <= -85 && average_signal_strength >-95)

    Signal quality=Bad

    Elseif (average_signal_strength <= -75 && average_signal_strength > -85)

    Signal quality=Average

    Elseif (average_signal_strength <=-65 && average_signal_strength > -75)

    Signal quality=Good

    Elseif (average_signal_strength <= -55 && average_signal_strength > -65)

    Signal quality=Very good

17. Write(“signal_measure.log”, date, time, phone status, average_ signal_strength)

18. If (sendSMS=auto && whenSMSsend=always || sendSMS=auto &&

whenSMSsend < bad)

sendSMS(average_signal_strength, signal quality, call_drop_information)

19. Write (signal quality)

20. If (GPS_coords available)

If(call_dropped=Normal)

Normal_dropped_calls=Normal_dropped_calls+1

Landmark_colour=green

Else

landmark_colour=red

hand-over_dropped_calls=hand-over_dropped_calls+1

Open(Nokia_map)

Plot(gps_coords, landmark)

21. If (total_calls=10)

Score_hand-over_dropped=

sum(hand-over_dropped_quality)/hand-over_dropped_calls

Score_normal_dropped=

sum(normal_dropped_quality)/normal_dropped_calls

score_successful_attempts=

(sum(hand-over_dropped_quality+sum(normal_dropped_quality))

/total_successful_attempts

22. If (call_attempts_successful< =2) Score_successful_call_rate = 1

    Elseif (call_attempts_successful< =3 && average_signal_strength >=4)

    Score_successful_call_rate=2

    Elseif (call_attempts_successful< =5 && average_signal_strength >=6)

    Score_successful_call_rate=3

    Elseif (call_attempts_successful< =7 && average_signal_strength >=8)

    Score_successful_call_rate=4

    Elseif (_attempts_successful< =9 && average_signal_strength >=10)

    Score_successful_call_rate=5

23. If (hand-over_success_calls< 40 per cent) Score_hand-over_success_calls_rate=1

    Elseif (hand-over_success_calls <40 per cent && hand-over_success_calls >60)

    Score_hand-over_success_calls_rate=2

    Elseif (hand-over_success_calls <60 per cent && hand-over_success_calls >70)

    Score_hand-over_success_calls_rate=3

    Elseif (hand-over_success_calls <70 per cent && hand-over_success_calls >80)

    Score_hand-over_success_calls_rate=4

    Elseif (hand-over_success_calls >80) Score_hand-over_success_calls_rate=5

24. Calculate

    average_call_quality=(score_successful_attempts+score_successful_call_rate+score_hand-over_success_calls_rate)/3

25. Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful, score_successful_attempts, normal_dropped_calls,score_normal_dropped, hand-over_dropped_calls,score_hand-over_dropped,score_successful_call_rate,score_hand-over_success_calls_rate, average_call_quality)

26. If (sendSMS=auto && whenSMSstat_send=always || sendSMS=auto && whenSMS_call_failed < 5) || whenSMS_hand-over_dropped < 2)

    sendSMS(num_calls_unsuccessful, num_calls_successful,

    num_of_calls_dropped_hand-over, num_normal_dropped)

27. If (log location=internal memory)

    save signalmeter.log to c:/data

    save calls_stats.log to c:/data

    else save signalmeter.log to e:/data

    save calls_stats.log to e:/data

28. If (sendSMS = Manual && want_to_send_SMS= yes)

    set(mobile_number)

    sendSMS(signal_s10gth, SignalQuality,call_drop_information)

29. End of programme

6.3 Bandwidth quality algorithm

A generic algorithm of bandwidth quality is presented below.

```
1.  Get the preferences for log_change, log_location, number_dowload
2.  If (start_download = true)
     save current time as Tb.
3.  While (start_download= true)
4.  Continue
5.  End of while
6.  Save current time as Te and file size Fs
7.  Calculate average speed S=Fs/(Te – Tb)
8.  Case S of
     <10: Score=1 (Very bad)
     >10 and < 50: Score=2 (Bad)
     >50 and <80: Score=3 (Average)
     >80 and < 100: Score= 4 (Good)
     > 100: Score=5 (Excellent)
9.  Write("bandwidth.log", S, score)
10. If (number_of_downloads=10)
11. Reset num_download= 0
12. Calculate S_average_10=(S1 + S2 + .... + S10) /10
13. Case S_average_10 of
     <10: Score=1 (Very bad)
     >10 and < 50: Score=2 (Bad)
     >50 and <80: Score=3 (Average)
     >80 and < 100: Score=4 (Good)
     > 100: Score=5 (Excellent)
14. Write("bandwidth_stats.log", S_average_10, Score)
```

15. Else number_of_downloads= number_of_downloads+1
16. Save number_of_downloads
17. End of else if
18. End of bandwidth_quality_measure

The application has been developed in Symbian c++. The details of classes used in the application are referred to in Appendix A. The log file data for one active call can be seen in Appendix B.

6.4 Signal meter modules

Below are the flow charts of individual processes of the complete signal meter system [60]. The system is divided into five individual processes, namely signal strength measurement, signal strength statistics, successful call rate statistics, plotting landmarks and SMS signal information.

6.4.1 Signal strength flow chart

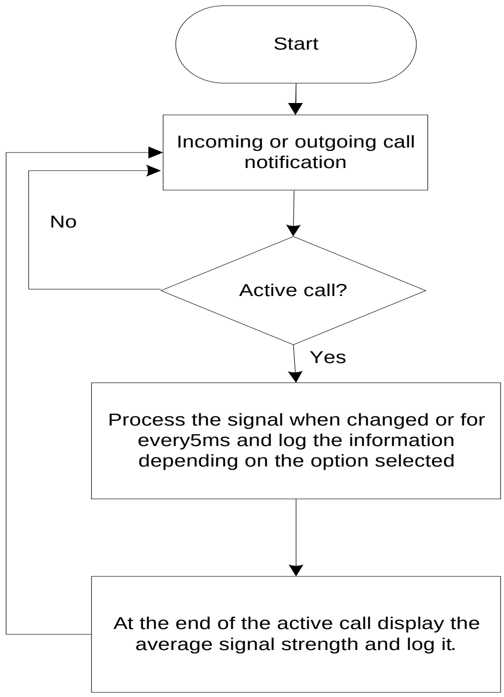


Figure 19: Signal strength measure flow chart

Figure 19 illustrates the flow of signal strength measurement for an active call. The signal strength of the active call is processed either every five milliseconds or for every change in strength. At the end of the call the information is displayed on the user interface screen of the system and the record can also be viewed later in the log file.

6.4.2 Plotting landmarks flow chart

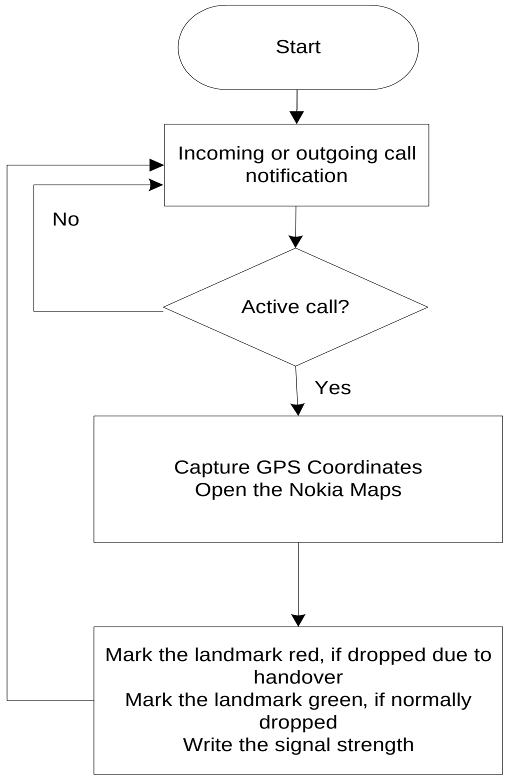


Figure 20: Plotting landmarks

Figure 20 illustrates how the GPS co-ordinates are captured in the system and signal information of the call at that particular location is plotted in the form of a landmark. The landmark is coloured red for calls dropped during the hand-over and green when calls are normally dropped.

6.4.3 Signal meter statistics flow chart

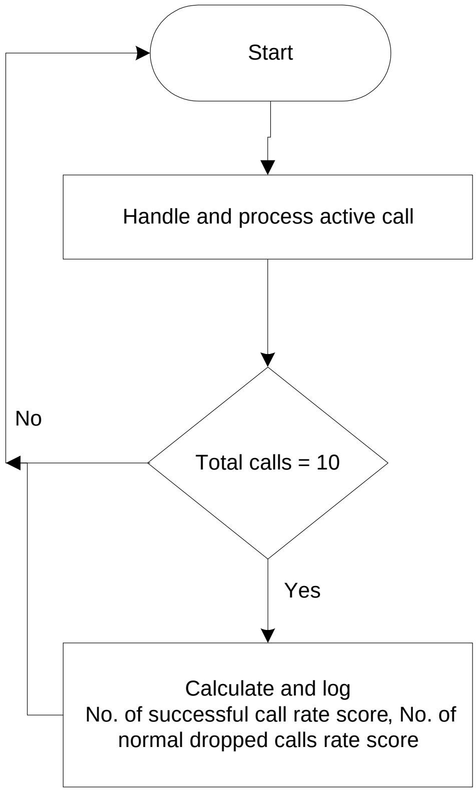


Figure 21: Signal meter statistics flow chart

Figure 21 illustrates the process flow for a bundle of the last 10 calls. The system records successful and unsuccessful call attempts, calls dropped due to hand-over and calls normally dropped. This is the most critical information for the analysis of signal quality. More on this is provided in Chapter 7.

6.4.4 SMS signal meter flow chart

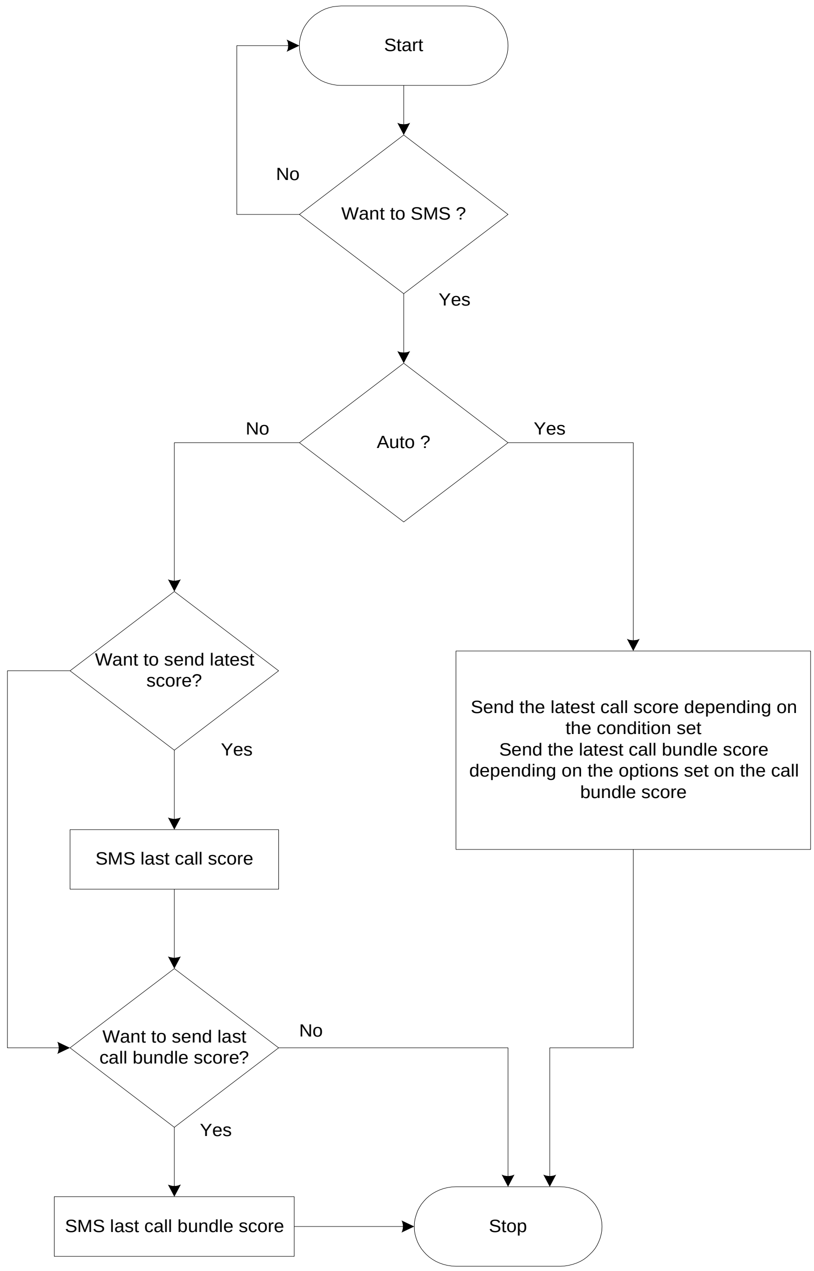


Figure 22: SMS signal meter flow chart

Figure 22 illustrates the process flow of SMS messages sent automatically and manually. The system can send signal strength information to a particular number. It is possible to select the mobile number to which the relevant SMS would be sent automatically at the end of the call. It is also possible to select whether to send the SMS at the end of every call, at the end of, for example, every tenth call or at the end of every call where call quality was poor. The call statistics can also be sent as an SMS.

6.4.5 Signal meter flow chart

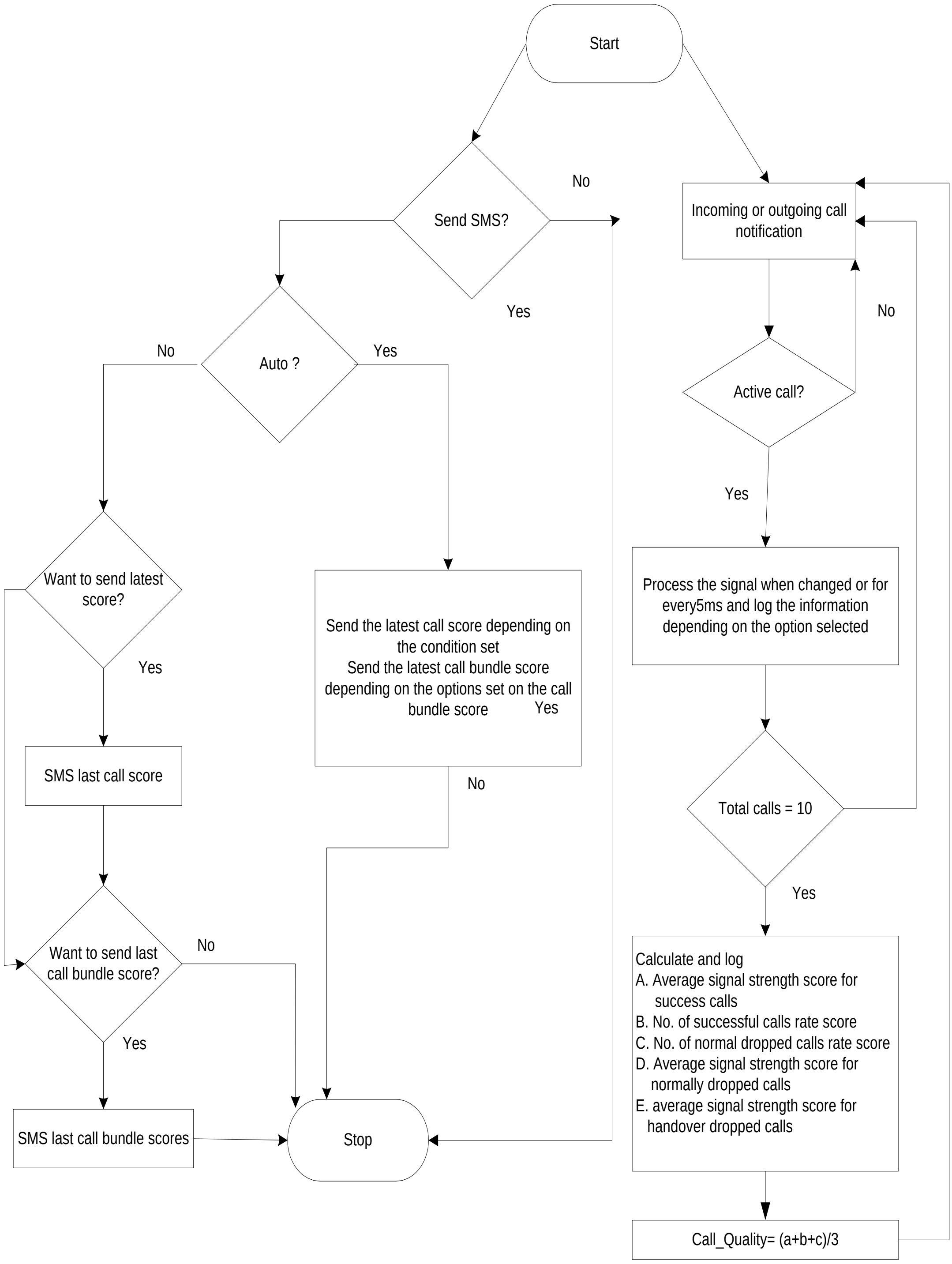


Figure 23: Signal meter flow chart

The signal meter flow chart is a consolidated flow chart of the sub-modules: signal strength calculation, signal meter statistics, landmark plotting and SMS signal meter.

6.5 Bandwidth meter flow chart

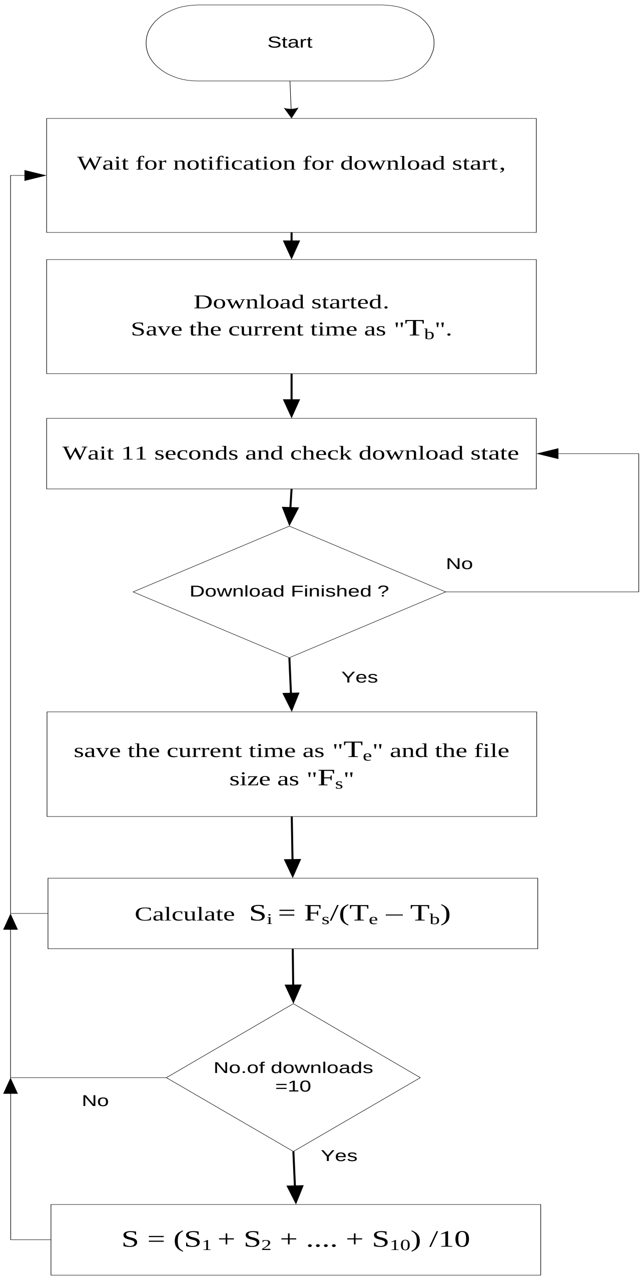


Figure 24: Bandwidth meter flow chart

Bandwidth meter is the quality meter for the GPRS sessions as experienced by the end-user. The bandwidth flow chart presents the process flow of the bandwidth quality calculation proposed and implemented. It helps user-groups and operators to evaluate bandwidth quality.

In the bandwidth meter flow chart above:

T_b: Start of file download

T_e: End time of the file download

F_s: File size

S_i: Average speed of download for the i^{th} file download

S: Total average speed for "n" file downloads

n: Total file downloads.

1. When a new download is initiated save the current time as "T_b".
2. When the download is finished save the current time as "T_e" and the file size as "F_s".
3. Calculate the average speed for this download as $S_i = F_s/(T_e - T_b)$.
4. Calculate the total average speed: if average speed is calculated for "n" downloads then total average speed is $S = (S_1 + S_2 + ... + S_n) /n$.

CHAPTER 7

Experiment and results

7.1 Sampling rate

Within the Symbian operating system there is provision for implementing a range of sampling intervals. In selecting a sampling interval it is reasonable to relate this to some other system feature. For example, the digital European cordless telecommunications (DECT) frame interval is 10 milliseconds, while at the opposite extreme is the frequency of gain adjustments in the universal mobile telecommunications system (UMTS) closed-loop control system, at every 0.67 milliseconds. Since the main subject of this research is the global system for mobile communications (GSM) system, where the frame duration is 4.615 milliseconds, an interval of 5.0 milliseconds has been chosen as a good approximation to the GSM frame interval and used in subsequent experiments.

7.2 Experiment and results

The results of the experiments are stored in log files as shown in section 6.3. One file contains the average signal strength of the call and the other the statistics of last 10 calls. The signal strength is measured every five milliseconds and stored in the log file along with the location code and time stamp. There is also provision for recording only the change in signal strength rather than recording every five milliseconds. The call log record is shown only when signal strength changes. The signal strength is categorized on a scale of 1-5, as mentioned in Chapter 3. The signal strength measured is depicted as a landmark on the Nokia map. The statistics of calls according to whether they are successful, unsuccessful because the network is busy or because of hand-over failure, normally dropped calls and so on, are also recorded for every 10 calls.

7.3 Signal information record [61]

2009/05/21 - 13:20:35 : : Current network info LocationAreaCode = 352 CellId = 12211

2009/05/21 - 13:20:36 : : Signal strength = 80 dBm, 7 bars

2009/05/21 - 13:20:58 : : Signal strength = 83 dBm, 7 bars

2009/05/21 - 13:20:59 : : Signal strength = 82 dBm, 7 bars

2009/05/21 - 13:21:07 : : Signal strength = 77 dBm, 7 bars

2009/05/21 - 13:21:12 : : Signal strength = 81 dBm, 7 bars

2009/05/21 - 13:21:44 : : Signal strength = 79 dBm, 7 bars

2009/05/21 – 13:21:46 : : Signal strength = 82 dBm, 7 bars

2009/05/21 - 13:21:47 : : Signal strength = 78 dBm, 7 bars

2009/05/21 - 13:21:49 : : Call drop observer -> Event : Call state is changed. Phone status: Idle

2009/05/21 - 13:21:49 : : Average signal strength is 80 dBm (average)

7.4 Sample call statistics

2009/05/24 - 07:45:33 : : 0 call attempts failed

2009/05/24 - 07:45:33 : : 10 call attempts successful :: Score: 3 (Average)

2009/05/24 - 07:45:33 : : 10 calls normally dropped :: Score: 3 (Average)

2009/05/24 - 07:45:33 : : ***********************

7.5 Landmarks on the map

Figure 25: Landmarks on the map

The Symbian operating system can send signal strength information and can be configured to set exactly when to send these details. Different options – such as if the signal strength is bad for particular call, if the number of unsuccessful attempts is less than five in a bundle of 10, if the number of dropped calls due to hand-over is less than two in a bundle of 10 – can be set. Apart from these options, the system can send an SMS whenever the user needs to send information on the signal strength of a recent call. The system also has the facility always to send an SMS, irrespective of the parameters mentioned above [62].

7.6 SM (signal meter) parameters comparison for operators

The experiment was carried out for a bundle of 10 calls, repeated 10 times in each network, and the average scores obtained. The experiment was therefore conducted for 100 calls for each network over a period of six months and at varying locations. The scores for the four parameters – successful call attempts, unsuccessful call attempts, normally dropped calls and calls dropped due to hand-over – were compared for two networks of Operator A and Operator B. The best scores of both operators are shown in Figure 26. The worst scores of Operator A and Operator B are shown in Figure 27 and Figure 28 respectively [63].

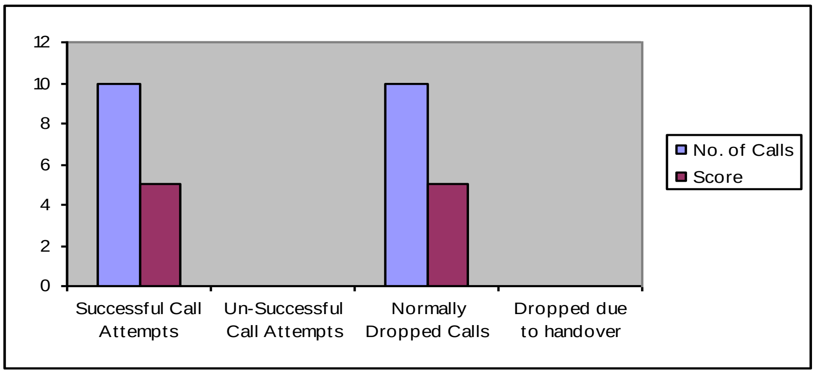


Figure 26: Operator A and Operator B best scores

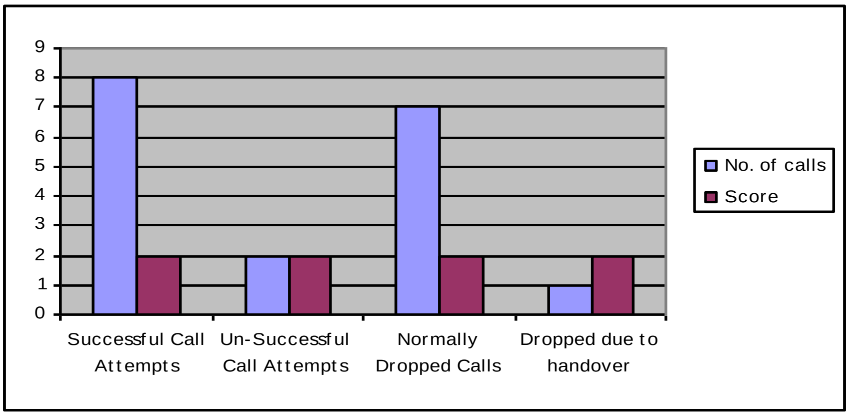


Figure 27: Operator A worst score

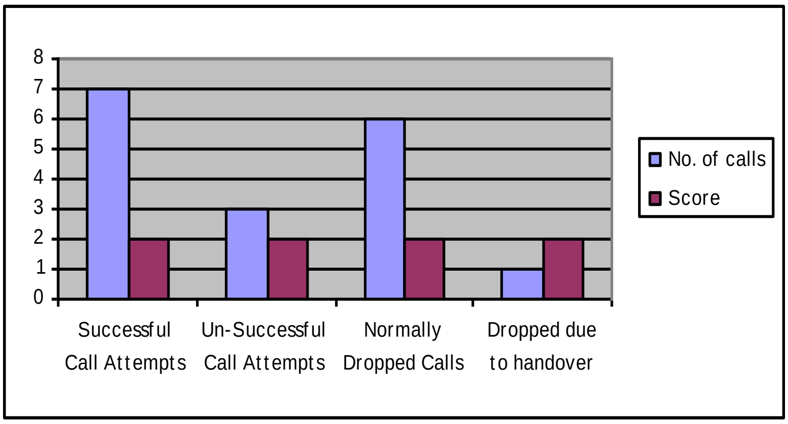


Figure 28: Operator B worst score

This experiment was performed to demonstrate one of the practical applications of the SM. It can be further fine tuned by the operators and user groups to suit their purposes and needs. Figures 26-28 demonstrate that network operators' best scores are the same, but their worst scores vary significantly. The variations presented are statistically significant. They can be interpreted by different user groups according to their requirements. For example, a user group can select the network that has the best rating in its area. And operators can improve their network quality through end-user benchmarking.

CHAPTER 8

QMeter® applications

8.1 Introduction

The proposed parameters to test the quality of calls are applicable at the levels of user, operator and regulator.

8.2 User layer

The proposed parameters can be used by the individual or by user groups such as companies and organizations to benchmark call and bandwidth quality before choosing a network. The user or user groups can use the various quality parameters or the aggregated call quality parameter, depending on their purpose and need.

The proposed and developed applications can be used on other platforms – such as Android, iPhone, Unix and Windows platforms for all mobile and smart-phone devices – and hence can cover a wide range of handsets.

8.3 Operator layer

The call quality parameters are equally important from the operator’s perspective. The extensive call quality statistics obtained from the end-user’s perspective at different locations can be used by the operator to enhance quality parameters at various locations according to the demand. The operator can also publish this information to provide transparency of its network quality parameters and coverage, which in turn can attract subscribers.

The operator can use the proposed network quality parameters to gauge the call quality and charge the customer based on the call or bandwidth quality perceived by the end-user. Tariff propositions could be proposed based on individual and combined call quality parameters. These charges would increase the credibility of the operator in the market and so attract subscribers. Such a service could differentiate the operator from its competitors, increase customer value and lengthen the lifetime of a subscription [64].

Call quality is widely used by carriers that transport the calls of telecom operators. From the end-user's perspective, however, call quality has not been taken very seriously until now. This is the reason for the surge in GSM gateways that resulted in interconnect revenue loss and, more importantly, a decline in call quality.

It is incumbent upon operators to monitor call quality from the end-user's perspective, in order to retain subscribers and reduce subscriber "churn". Operators worry not only about call quality and interconnect revenue loss, but also network connectivity issues in areas where GSM gateways are prevalent. Finally, operators must monitor their network quality to maintain their key performance indicators.

Bandwidth quality as experienced by the end-user is equally important in helping operators to reduce churn. General packet radio service (GPRS) is becoming the most widely used technology on mobile after voice. Increases in the urban population mean that operators have to adapt according to usage, with some parts of the urban area using less GPRS and others comparatively more.

The proposed parameters were developed keeping in mind all the parameters that influence call and bandwidth quality as experienced by the end-user. The system can also be used to benchmark the network as a key performance indicator by using the landmarks plotted on the map.

An alternative tariff proposition could be made towards a particular subscriber segment to satisfy its demand for quality voice and data services. This would increase the credibility of the operator in the market. Scores for call and bandwidth quality revealed in the research (the subjective scores were rated by third party individuals and correlation was made by the author of this thesis) correspond closely with people's subjective scores. This demonstrates and increases the credibility of the parameters proposed.

By adopting the proposed method, operators would be able dramatically to increase their addressable target issues related to call and bandwidth quality in market. This would help them to improve their credibility and accelerate revenue generation by providing quality of service in the market.

The research addresses quality related issues of voice and data of mobile operators with respect to end-user. The mobile operator can use the results in different ways:

- Based on the parameters, the operator can analyze quality drop. This may be due to illegal GSM gateways operating in specific regions. The operator can take appropriate action to disconnect, based on the set quality threshold, or fix the issues related to its key performance indicators.
- Information on the data speed of sessions can be collected at the central database, analyzed and used to enhance the service. This would enable the operator to charge the subscriber according to the quality parameters that he or she perceives.
- The operator can collect the quality results for voice calls and data sessions for VIP subscribers, analyze the quality scores and apply the relevant tariff proposition [65].

8.4 Regulator layer

Call quality parameters can also be used by telecom authorities to check whether mobile network operators are meeting their licence criteria of network quality from the end-user's perspective. The quality parameters can also be used as a consumer protection tool to ensure that tariffs correlate with call quality [65].

Regulators can set the quality threshold to ensure that operators are meeting certain quality parameters from the end-user's perspective. This is very much needed, because of growing discontent among users over the quality of voice and data. Sometimes users get billed for calls they could not make or for calls that were quickly disconnected because of connectivity problems. These kinds of problem are arising more frequently because of the huge rise in the number of GSM gateways. For the last couple of years, operators have been struggling to gain control over illegal GSM gateways by applying various procedures and disconnecting them. But the lack of proper procedures in off net calls means that the operators are unable to deal with these illegal GSM gateways in off net networks. Regulators can insist that all operators meet required quality standards; this, in turn, would automatically reduce the number of illegal GSM gateways.

CHAPTER 9

Proposed charging rate versus quality

9. 1 Charging rate versus average signal strength of successful calls [66]

Table 8 shows the new charging rate proposed based on the average signal strength of successful calls in a bundle of 10. The formula used for charging is: X*n*0.75 for the average score, X*n*0.5 for the bad score and no charge for the extremely bad score. X is the normal charging rate per minute of the call and n is the number of called minutes in the bundle of 10.

Average signal strength of successful calls score	**Charge**
5	X*n
4	X*n
3	X*n*0.75
2	X*n*0.5
1	No charge

Table 8: Proposed charging rate versus average signal strength of successful calls

9.2 Charging rate versus successful call attempts

Table 9 shows the new charging rate proposed based on successful call attempts in a bundle of 10. The formula used for charging is: X*n*0.75 for the average score, X*n*0.5 for the bad score and no charge for the extremely bad score. X is the normal charging rate per minute of the call and n is the number of called minutes in all the successful call attempts in the bundle of 10.

Successful call attempts score	**Charge**
5 (Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 9: Proposed charging rate versus successful call attempts

9.3 Charging rate versus normally dropped rate

Table 10 is the new charging rate proposed based on the normally dropped rate in a bundle of 10. The formula used for charging is: X*n*0.75 for the average score, X*n*0.5 for the bad score and no charge for the extremely bad score. X is the normal charging rate per minute of the call and n is the number of called minutes in all the normally dropped calls in the bundle of 10.

Normally dropped rate score	**Charge**
5 (Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 10: Proposed charging rate versus normally dropped rate

9.4 Charging rate versus total call quality

Table 11 shows the new charging rate proposed based on total call quality of calls in a bundle of 10. The call quality score covers the successful calls rate, the average signal strength of successful call attempts and the normally dropped rate. The formula used for charging is: X*n*0.75 for the average score, X*n*0.5 for the bad score and no charge for the extremely bad score. X is the normal charging rate per minute of the call and n is the number of called minutes in the bundle of 10.

Call quality	**Charge**
5 (Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 11: Proposed charging rate versus total call quality

9.5 Charging rate versus total bandwidth quality

Table 12 shows the new charging rate proposed, based on the average bandwidth per 10 downloads. If X is the charging rate for the downloaded content, the tariff based on the bandwidth quality can be applied as follows:

Bandwidth quality score	**Charge**
5 (Very good)	X
4 (Good)	X
3 (Average)	X*0.75
2 (Bad)	X*0.5
1 (Very bad)	No charge

Table 12: Proposed charging rate versus bandwidth quality

CHAPTER 10

Conclusions

10.1 Introduction

These conclusions consider what has been achieved to date and what is perceived as further development of the research in the light of anticipated innovations in the evolution of both system and operating system design.

10.2 Current status

Call quality is a parameter widely used by carriers that transport the calls of telecom operators. From the end-to-end-user's perspective, however, call quality has not been taken very seriously until recently. This is the reason for the surge in access by GSM gateways. This has resulted in interconnect revenue loss for operators and, more importantly, degradation of call quality.

Similarly, bandwidth quality as measured by accessible user bandwidth and as experienced by end-users is equally important for operators in their strategies to retain their customer base and to reduce churn. GPRS (general packet radio service) is becoming the most widely used technology on mobile after voice. Because of demographic changes, mostly in urban areas, operators have to adopt according to usage. This is commented on further later in this chapter.

Now it has become important for all incumbent operators to monitor call quality from the end-user's perspective in order to ensure that subscribers are satisfied with the services they receive and that these are commensurate with the charges levied. Apart from call quality and interconnect revenue loss, operators are worried about network connectivity in areas where GSM gateways are prevalent. Finally, operators have to monitor their network quality to maintain their key performance indicators.

Analysis of speech quality is a major research topic for current wireless networks. Voice quality is a major issue since it is in real time. Consumers will tolerate delay, noise and jitter only to certain levels. Traditionally, a user's perception of speech quality is measured offline using subjective listening tests. Such tests are, however, slow and costly.

In this research, a Symbian based method for measuring speech quality in a mobile cellular telecommunication network is presented. The method adopts a non-traditional approach and can be used as a relative index for assessing call quality. The method involves investigation of metrics which are used to detect the call, obtaining signal strength information and refreshing the value every five milliseconds. The call quality is derived based on parameters such as the number of call attempts successful, the number normally dropped, the number of hand-over dropped calls and the average signal strength. Call quality is derived based on call statistics. Issues such as visualizing and escalating call quality information are also considered as a part of the research.

The QMeter® tool in this research measures signal and bandwidth quality as experienced by the end-user. The research concentrates on the subscriber and his or her demand for quality services. It proposes varying tariffs according to call and bandwidth quality parameters. QMeter® has a built-in tool that uses maps to landmark call quality. QMeter® can also be used to benchmark the network as a key performance indicator. By adopting QMeter®, mobile network operators can dramatically increase their addressable target issues related to call and bandwidth quality, and improve their credibility and revenues by providing high quality service.

Network operators are increasingly considering the Android operating system as the basis for their new service offerings. While this research focuses on Symbian, it is evident that a similar strategy could be applied to Android. This is considered in more detail in 10.4.

Telecom authorities can also use QMeter® to regulate and evaluate mobile operators. They can do this by regularly checking the network to ascertain whether mobile operators are meeting their licence criteria of network quality from the end-user's perspective. Further, QMeter® can be used as a consumer protection tool to ensure that tariffs correlate with call quality.

By adopting QMeter®, mobile network operators will be able dramatically to increase their addressable target issues related to call and bandwidth quality and improve their credibility and revenue generation in the marketplace by providing evidence of the quality of their service.

In future, mobile operators could extend this research to compare their benchmarking with user benchmarking. To address quality of service from the subscriber's point of view, a varying tariff based on call and bandwidth quality parameters could further be improved and developed as a part of QMeter®.

At the operator end, information on call quality and data speed of sessions could be stored initially in the central database for VIP subscribers. Quality scores could be analyzed and different tariffs applied. This could later be applied to different subscriber segments, once it has successfully been applied to the small number of VIP subscribers.

10.3 Directions in system design

There are two aspects to the ways in which advances in system design are relevant to the research which is the subject of this thesis: the evolution of the "femto cell" approach to service delivery and the emergence of "cognitive radio".

10.3.1 The femto cell approach

The conventional approach to service delivery has followed a strategy of accommodating growing numbers of users by progressively reducing cell sizes to achieve more intensive reuse of the fixed spectrum resources allocated to mobile communications. Additionally, users are continuing to demand support for the larger volumes of data traffic associated with downloading of real time video and other entertainment material. The result has been that cell sizes have ultimately reduced to the point at which a single building forms a femto cell. The question then arises as to how femto cells can co-exist with micro and macro cells. There are various possibilities.

One solution is for an operator managing a conventional micro/macro cellular environment to add femto cells into the management process. However, increasingly there is a perception that individual users should be able to create their own femto cells without co-ordination with an existing operator. But since such femto cells must share the same radio spectrum as surrounding micro/macro cells there are potential problems. The femto cells may create interference in the surrounding cells with impact on the effective bandwidth available to the wider user community which, in turn, has consequences for the speech quality that users experience. A further consideration is the possibility of hand-over from a femto cell into surrounding cells and *vice-versa*. If this were to happen, how would a charging regime be implemented and how would individual mobiles report the quality of speech which users experience?

10.3.2 Cognitive radio

In contrast to the femto cell situation, the concept of cognitive radio relates to the possibility of expanding both user population and service capability by using unlicensed windows in the radio spectrum.

Internationally, interest has focused on the parts of the broadcast television spectrum generally known as “white spaces”, where services can operate without causing interference with broadcast services. This opportunity has been under consideration for an extended period, possibly up to 10 years. In Europe, the electronic communications committee of the European Conference of Postal and Telecommunications Administrations (CEPT) has worked on the problem, while in the UK the independent regulator and competition authority for the communication industries, Ofcom, is also engaged with the issue. Similarly, in USA, the Federal Communications Commission (FCC) has identified the 2360 – 2400MHz band as a host for such applications as “smart grid”, remote meter reading and “medical body area networks” (MBANs). Additionally, there is an expectation that spectrum will also be used to support further high data-rate mobile services.

Inevitably, exploitation of this spectrum resource requires a new level of sophistication in both the capability of mobile terminals and the support for mobiles in the fixed infrastructure. In the context of research focus in speech quality measurement, there is a significant area of further work associated with cognitive radio. An overview of the current status of research into cognitive radio is presented in [67].

10.4 Operating system developments

There was mention above that the Symbian operating system (OS) has been the OS of choice by most mobile handset manufacturers in the last decade, but as with most software associated with hardware developments, the lifetime of an OS is limited. New applications are being created in great numbers as the capabilities of mobile handsets expand and new operating systems are being developed to exploit this capability.

The dominant OS currently being widely adopted is the Android OS, originated by a company based in California and subsequently acquired by Google. Android Inc was founded in 2003 and acquired by Google in 2005. Subsequently, Google undertook to facilitate the development of new applications using the Android platform and at the end of 2007 an "open handset alliance" was established bringing together major equipment manufacturers and operators committed to open access to the platform.

The success of this initiative is evidenced by the numbers of applications (apps) now available, currently estimated at 300,000. In terms of the number of devices worldwide now using the Android platform, a recent (October 2011) estimate claims 190 million devices: "smart-phones" and "tablet computers" and specialist user equipment [68].

In relation to the current research, it is clear that the achievements reported in using the Symbian OS can be readily developed into similar applications in Android. The Android kernel is Linux based with peripherals originally in C. Apps developers write in a customized version of Java. Given the pace of developments in this technology, it is considered desirable that an Android version of the Symbian functionality reported here should be available by late 2012.

10.5 Final remarks

This research has demonstrated that, by using the capabilities of mobile handset functionality, coupled with that of the Symbian operating system, new opportunities exist for subscribers. Network operators and service providers can now deliver attractive and cost effective subscriber contracts in which charges are determined by quality of service. While the emphasis has been on the Symbian operating system, extension to the Android OS is readily achievable.

Appendix A

Signal meter classes and their description

This appendix describes the various signal meter classes: Class CSignalStrength; Class CConfig; Class CLogger; Class ClineListener; Class CMainView; Class CMainViewContainer; Class CPhoneShell; Class CPeferencesDlg; Class CTimeOutTimer; Class CNetworkListener; Class CBluetoothEngine; Class CActivePositioner; and Class TNetworkInfo [69].

Class CSignalStrength

Description:

This class gets the phone's current signal strength

Class members:

void GetSignalStrength(TRequestStatus& aReqStatus, TInt32& aSignalStrength, TInt8& aBar)

Description:

This function member gets the phone's current signal strength

Parameters:

TRequestStatus &aReqStatus - On return, KErrNone if successful, KErrNotSupported if signal strength information is not supported by the phone, and KErrNotFound if no signal strength information is currently available

TInt32 &aSignalStrength - On completion, the signal strength in dBm, or 0 if the TSY does not wish to provide this information

TInt8 &aBar - On completion, the number of bars the phone should display, or -1 if the TSY does not wish to provide this information

Class CConfig

Description:

This class provides read/write operations from the SignalMeter.ini file. We use this .ini file for storing preferences such as signal strength “refresh interval” and the location of the “store location” log file

Class members:

void CConfig::SetLogSettingsL(const TInt aRefreshInterval , const TInt aSaveLocation)

Description:

This function sets the application settings

Parameters:

TInt aRefreshInterval -Possible values (0 - Every 5 ms , 1 - When changed)

TInt aSaveLocation - Possible values (0 - Internal memory, 1 - Memory card)

void CConfig::GetLogSettingsL(TInt& aRefreshInterval ,TInt& aSaveLocation)

Description:

This function gets the application settings

Parameters:

Tint& aRefreshInterval - Possible values (0 - Every 5 ms , 1 - When changed)

Tint& aSaveLocation - Possible values (0 - Internal memory, 1 - Memory card)

Class CLogger

Description:

This class provides written signal strength information during a call .We write this information in the SignalMeter.log file. The location of this file depends on the “store location” value from CPreferencesDlg. The possible locations are c:\\data\\SignalMeter.log, and e:\\data\\SignalMeter.log

Class members:

void Logger::Log(const TDesC& aValue , const TInt aDrive)

Description:

This function provides a tool for saving the results from measuring signal strength and the average signal strength at the end of each call

Parameters:

const TDesC& aValue – Text to write in to a log file

const TInt aDrive – Drive location where the log file will be saved . Possible values (0 - Internal memory, 1 - Memory card)

Class ClineListener

Description:

With this class we "listen" for changes in the phone "line status"

Class members:

void RMobileLine::NotifyStatusChange(TRequestStatus& aStatus,RCall::TStatus& aLineStatus)

Description:

Provides notification about a change in the line status

Parameters:

TRequestStatus& aStatus - A variable that indicates the completion status of the request

RCall::TStatus& aLineStatus - On request completion, contains the new line status

The possible results are:

EstatusUnknown	The call status is not known
EstatusIdle	The call is idle
EstatusDialling	The call is dialling
EstatusRinging	The call is ringing (an incoming, unanswered call)
EstatusAnswering	The call is being answered

EstatusConnecting		The call is connecting (immediate call establishment, without dialling)
EstatusConnected	-	The call is connected and active
EstatusHangingUp		The call is being terminated

Class CMainView

Description:

This is the main window when the application is started. Here we show texts such as: “Phone status” and “Average signal strength”

Class members:

void CMainView::HandleCommandL(TInt aCommand)

Description:

This function handles user commands, such as “Preferences”, “About” and “Exit”

Class CMainViewContainer

Description:

This class provides a controls drawing from the CMainView class

Class members:

void CMainViewContainer ::CreateLabelL(CEikLabel** aLabel, const CFont& aFont , const TDesC& aTextToShow)

Description:

This function provides for creating a text label control

Parameters:

CEikLabel** aLabel - A variable that represents a pointer to a label that must be created

const CFont& aFont - A variable that specifies the font that will be displayed for text

const TDesC& aTextToShow – Text that must be displayed in this label control

void CMainViewContainer ::SetLabelCtrlPosition(CEikLabel* aLabel, TUint aTopCo-ordinate)

Description:

Sets the label control position on the screen

Parameters:

CEikLabel* aLabel – Pointer to label control than must be moved

TUint aTopCo-ordinate – A top Y co-ordinate of the label control rect

void CMainViewContainer ::RefreshPhoneStatusL(const TDesC& aBufMessageToShow)

Description:

This function displays a new call status received from RMobileLine::NotifyStatusChange

Parameters:

const TDesC& aBufMessageToShow – Text to show in the label control

void CMainViewContainer::SetAverageSignalStrengthL(const TDesC& aBufMessageToShow)

Description:

This function displays the average signal strength when the call is finished

Parameters:

const TDesC& aBufMessageToShow – Text to show in the label control

Class CPhoneShell

Description:

This class provides the main application functionality. From here we make:

- start line listening (CLineListener)

- get signal strength during "Refresh interval" (CTimeOutTimer , CSignalStrength)

- calculate average signal strength (function CalculateAndWriteAverageSignalStrengthL()) set new values of "Refresh interval", "Store location", when they were changed from CPreferencesDlg

Class members:

void CPhoneShell ::ConnectToPhoneServerL()

Description:

Connects the client to the ETel Server. It must be used before any other functions during a telephony session

Parameters:

None

void CPhoneShell ::DisconnectFromPhoneServer()

Description:

Disconnects the client from the ETel Server

Parameters:

None

void CPhoneShell ::PhoneIsConnected()

Description:

This function notifies the user when he or she has a phone call

Parameters:

None

void CPhoneShell ::PhoneIsDisconnected()

Description:

This function notifies the user when the phone call is ended

Parameters:

None

void CPhoneShell CalculateAndWriteAverageSignalStrengthL()

Description:

Calculates and writes into log file the average signal strength for the previous phone call

Parameters:

None

void CPhoneShell ::SetNewSettings (const TInt aRefreshInterval, const TInt aSaveLocation)

Description:

The function is called upon when users have changes in preferences ("Refresh interval" or "Store location")

Parameters:

Tint& aRefreshInterval - Possible values (0 - Every 5 ms , 1 - When changed)

Tint& aSaveLocation - Possible values (0 - Internal memory, 1 - Memory card)

void CPhoneShell::GetCurrentNetworkCompleted (const TInt aErrorCode, const TNetworkInfo& aCurrentNetwork)

Description:

This function is called upon when the process of detecting the current network to which the phone is connected at the particular moment has finished

Parameters:

Tint aErrorCode - Possible values (KErrNone if successful, otherwise a code of the error)

TNetworkInfo& aCurrentNetwork – This parameter contains information about the current network. See TNetworkInfo

void CPhoneShell::NetworkHasBeenChangedIndependently (const TInt aErrorCode, const TNetworkInfo& aCurrentNetwork)

Description:

This function notifies the user when the network to which the phone is connected at a particular moment has changed. Also it writes the information about the new network into a log file

Parameters:

Tint aErrorCode - Possible values (KErrNone if successful ,otherwise a code of the error)

TNetworkInfo& aCurrentNetwork – This parameter contains a information about the new current network . See class TNetworkInfo

void CPhoneShell::NotifyForGPSCo-ordinates (const TPositionInfo aPositionInfo)

Description:

This function is called upon when the GPS co-ordinates are changed. The function also writes these co-ordinates into a log file

Parameters:

TPositionInfo aPositionInfo - Contains the new GPS co-ordinates

void CPhoneShell::StartNetworkLis10erL()

Description:

This function starts the process which “listens” for network status changes

Parameters:

None

void CPhoneShell::StartSignalStreingthLis10erL()

Description:

This function starts the process of checking for changes in the power of the signal. This is done in a time interval (5000 microseconds)

Parameters:

None

void CPhoneShell::StartActivePositionerL()

Description:

This function starts the process which “listens” for GPS co-ordinates changes. See CActivePositioner

Parameters:

None

void CPhoneShell::CheckForCellChangeL (const TUint aCurrentCellId)

Description:

The function checks and writes in a file, if the current cell is different from the last saved cell to which the phone was connected

Parameters:

const TUint aCurrentCellId – It consists the number of the current cell to which the phone is connected

void CPhoneShell::CheckForCallStatusChangeL (const TDesC& aBufPhoneStatusMessage)

Description:

The function checks and writes in a file, if there is a change in the current phone call status

Parameters:

const TDesC& aBufPhoneStatusMessage – It contains the current phone call status. Possible values:

- Phone status: Unknown
- Phone status: Idle
- Phone status: Dialling
- Phone status: Ringing
- Phone status: Answering
- Phone status: Connecting
- Phone status: Connected
- Phone status: Disconnecting
- Phone status: Disconnecting with inband
- Phone status: Reconnect pending
- Phone status: Hold
- Phone status: Waiting alternating call switch
- Phone status: Transferring
- Phone status: Transfer alerting

void CPhoneShell::CheckForCallDropL()

Description:

The function checks and writes in a file if the current phone call has dropped because of a bad signal

Parameters:

None

void CPhoneShell::ChangeNetworkLis10MetodL(const TBool aNotifyWhenChanged)

Description:

The function defines the way in which it checks for changes in the power of the signal in the current network

Parameters:

TBool aNotifyWhenChanged : If it is TRUE, in the log file it writes the power of the signal, only if there is a change in comparison to the last value

If it is FALSE, the power of the signal is written at an interval of time, no matter if there is a change or not

Class CPeferencesDlg

Description:

This class provides the user interface for changing the values of “Refresh interval” and “Store location”

Class CTimeOutTimer

Description:

This class provides a time interval during which the user must get signal strength

Class members:

void CTimeOutTimer ::After(TTimeIntervalMilliSeconds32 aInterval)

Description:

Requests an event after an interval. This timer completes after the specified number of milliseconds

Parameters:

TTimeIntervalMilliSeconds32 aInterval - Interval after which the event is to occur, in milliseconds

void CTimeOutTimer::After(TTimeIntervalSeconds aSeconds)

Description:

Requests an event after an interval. This timer completes after the specified number of seconds

Parameters:

TTimeIntervalSeconds aSeconds - Interval after which the event is to occur, in seconds

Class CNetworkListener

Description:

This class notifies when there is a change of the current network to which the phone is connected

Class members:

void CNetworkListener::StartGetCurrentNetwork()

Description:

This function starts the asynchronical process, which checks the network to which the phone is connected at that particular time

Parameters:

None

void CNetworkListener ::CurrentNetworkWasChangedIndependently()

Description:

This function notifies if there is a change of the current network, i.e. for some reason the phones got disconnected and then connected to another network

Parameters:

TInt aErrorCode – KErrNone if successful, else value indicating error situation.

void CNetworkListener::ListenForNetworkChange()

Description:

This function starts the process which "listens" for network status changes

Parameters:

None.

Class CBluetoothEngine

Description:

This class ensures the connection between the phone and an external GPS through Bluetooth connection

Class members:

TInt CBluetoothEngine::SetPowerState (TBool aState)

Description:

Setter for BT MCM power mode: With this function power is turned on/off

Parameters:

TBool aState - If true, the Bluetooth is turned on

return TInt – Kerr – None if OK, or else value indicating error

TInt CBluetoothEngine :: GetPowerState(TBool& aState)

Description:

This function returns the power state of the Bluetooth (on/off)

Parameters:

TBool aState - If true, the Bluetooth is turned on

On return TInt – Kerr – None if OK, or else value indicating error

Class CActivePositioner

Description:

This function is used to make the primary connection to the location server. The RPositionServer class can also be used to discover what position technology “modules” are available. This class is used to obtain the current GPS co-ordinates

Class TNetworkInfo

Description:

A structure in which the information about the current network is recorded

Parameters:

TBuf<4> iCountryCode; - Code of the country in which the network exists and to which the phone is currently connected

TBuf<8> iNetworkId – Operator's code to whose network is the phone is currently connected

TUint iLocationAreaCode –Location area code

TUint iCellId – Identifier of the cell of the current network

TBool iConnected - This shows if the phone is connected to the current network

Appendix B

Signal meter log for every change in signal strength of an active call

This appendix presents the signal meter log for every change in the signal strength of an active call [70].

2009/12/30 - 18:57:10: : Phone status: Ringing

2009/12/30 - 18:57:34: : Phone status: Answering

2009/12/30 - 18:57:35: : Phone status: Connected

2009/12/30 - 18:57:36: : Current network info LocationAreaCode = 3461 CellId = 6951401

2009/12/30 - 18:57:36: : Current network info LocationAreaCode = -2139943553 CellId = 6293752

2009/12/30 - 18:57:36: : Call drop observer -> Event : The current cell is changed

2009/12/30 - 18:57:36: : Current network info LocationAreaCode = 3461 CellId = 6951401

2009/12/30 - 18:57:37: : Call drop observer -> Event : The current cell is changed

2009/12/30 - 18:57:37: : Signal strength=100 dBm, 5 bars

2009/12/30 - 18:57:40: : Signal strength=96 dBm, 5 bars

2009/12/30 - 18:57:41: : Signal strength=86 dBm, 7 bars

2009/12/30 - 18:57:41: : Signal strength=83 dBm, 7 bars

2009/12/30 - 18:57:42: : Signal strength=85 dBm, 7 bars

2009/12/30 - 18:57:42: : Signal strength=84 dBm, 7 bars

2009/12/30 - 18:57:45: : Signal strength=82 dBm, 7 bars

2009/12/30 - 18:57:48: : Signal strength=83 dBm, 7 bars

2009/12/30 - 18:57:51: : Signal strength=81 dBm, 7 bars

2009/12/30 - 18:57:54: : Signal strength=82 dBm, 7 bars

2009/12/30 - 18:57:59: : Signal strength=81 dBm, 7 bars

2009/12/30 - 18:58:02: : Signal strength=72 dBm, 7 bars

2009/12/30 - 18:58:10: : Signal strength=81 dBm, 7 bars

2009/12/30 - 18:58:11: : Signal strength=76 dBm, 7 bars

2009/12/30 - 18:58:25: : Signal strength=82 dBm, 7 bars

2009/12/30 - 18:58:26: : Signal strength=81 dBm, 7 bars

2009/12/30 - 18:58:27: : Signal strength=80 dBm, 7 bars

2009/12/30 - 18:58:30: : Signal strength=78 dBm, 7 bars

2009/12/30 - 18:58:30: : Signal strength=80 dBm, 7 bars

2009/12/30 - 18:58:31: : Signal strength=82 dBm, 7 bars

2009/12/30 - 18:58:38: : Signal strength=79 dBm, 7 bars

2009/12/30 - 18:58:38: : Signal strength=82 dBm, 7 bars

2009/12/30 - 18:58:39: : Signal strength=83 dBm, 7 bars

2009/12/30 - 18:58:49: : Signal strength=81 dBm, 7 bars

2009/12/30 - 18:58:54: : Signal strength=80 dBm, 7 bars

2009/12/30 - 18:58:57: : Signal strength=83 dBm, 7 bars

2009/12/30 - 18:59:01: : Signal strength=79 dBm, 7 bars

2009/12/30 - 18:59:05: : Signal strength=83 dBm, 7 bars

2009/12/30 - 18:59:07: : Call drop observer -> Event : Call state is changed. Phone status: Idle

2009/12/30 - 18:59:08: : Average signal strength is 82 dBm (Average)

Appendix C

Previously published work by the thesis author

C1 Journal articles

From the International Journal of Computer Networks & Communications, Wireless & Mobile Networks, *June 2011.*

QMeter Tools for Quality Measurement in Telecommunication Network

Akram Aburas[1] and Prof. Khalid Al-Mashouq [2]

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[2]Electrical Engineering Department, King Saud University, Riyadh, Saudi Arabia

1. Introduction

Call quality is usually measured by carriers on parameters such as ASR (answer seizure ratio), PDD (post dial delay), NER (network efficiency ratio), the number of calls for which these parameters have been analyzed and successful calls. Mobile operators use the threshold to filter the carriers not meeting their quality threshold. To address the issue of quality from the end-user's perspective, operators need additional tools on the subscriber handsets integrated into their network. Call quality has not been taken very seriously until now and this is the reason for the surge in GSM gateways which resulted in interconnect revenue loss and more importantly poor call quality.

Traditionally speech quality measurement techniques have used the subjective listening test called mean opinion score (MOS). It is based on human perceived speech quality, on a scale of 1 to 5, where 1 is the lowest perceived quality and 5 is the highest. Subjective listening tests are expensive, time consuming and tedious so, currently, most systems use an objective evaluation of speech quality using mobile computing techniques. Objective testing systems use automated speech quality measurement techniques. Three well known objective measurement techniques are perceptual speech quality measure (PSQM), the perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ).

It has become important for all mobile operators to monitor call quality from the end-user's perspective to retain subscribers and reduce churn. Apart from call quality and interconnect revenue loss, operators are worried about network connectivity issues in the areas where GSM gateways are prevalent. Call quality can be monitored in areas that are difficult for monitoring to take place, like residential areas. Finally, operators have to monitor their network quality to maintain their key performance indicators.

Conventional speech quality measurement involves lots of resources and is tedious. An alternative method is therefore needed to address quality from end-user's perspective. It should be automatically computed on the subscriber handset and available to the operator. At the same time, the results must be comparable with conventional subjective scores. The proposed QMeter® addresses these issues and, additionally, puts forward dynamic tariff propositions which will enhance the credibility of the operator.

Bandwidth quality as experienced by the end-user is equally important for operators to reduce the churn. Data users are increasing exponentially. Due to changes in demographics, mostly in urban areas, operators have to adapt according to usage.

QMeter® is a set of tools for signal and bandwidth measurement that are developed bearing in mind all the parameters that influence call and bandwidth quality experienced by the end-user. The system can also be used to benchmark the network as a key performance indicator. QMeter® has a built-in tool that uses maps to landmark call quality.

Our research shows that QMeter® and subjective scores are closely correlated – a fact that demonstrates and increases the credibility of QMeter®.

By adopting QMeter®, mobile telecommunication operators can dramatically increase their addressable target issues related to call and bandwidth quality and improve credibility and revenue generation by providing a higher quality of service.

2 Related work

Objective speech quality measurement techniques are mostly based on the input-output approach [1]. In input-output, objective measurement techniques basically work by measuring the distortion between the input and the output signal. The input signal would be a reference signal and output signal a received signal.

Input-output based speech quality assessment in objective speech quality measurement gave good correlations, with reaches up to 99 per cent in some cases [2]. Estimating speech quality without the presence of an input signal or reference signal is latest area of research. Objective measurement is basically achieved by correlating the results with the subjective quality measure.

Estimating speech quality without the presence of an input signal or reference signal is the latest area of research. Jin Liang and R. Kubichek [3] published a paper on output based objective speech quality using perceptually based parameters as features. The results demonstrated 90 per cent correlation. R. Kubichek and Chiyi Jin [4] used the vector quantization method, which achieved 83 per cent correlation.

An output based speech quality measurement technique using the visual effect of a spectrogram is proposed in [5]. An output based speech quality evaluation algorithm based on characterizing the statistical properties of speech spectral density distribution in the temporal and perceptual domains is proposed in [6]. The correlations achieved with subjective quality scores were 0.897 and 0.824 for the training data and testing data set respectively.

A time-delay multilayer neural network model for measuring output based speech quality was proposed by Khalid Al-Mashouq and Mohammed Al-Shaye in [7]. The correlation achieved for speaker and text was 0.87.

In this paper we present our work for determining call quality parameters such as average signal strength, successful call rate and successful hand-over rate with respect to signal strength and successful rate. Then final call quality is computed from the extracted parameters.

This research is a continuation of the work in [8-14]. A basic bandwidth quality measurement is proposed which can be used by both the operator and the user to evaluate the bandwidth quality of a particular operator.

3. QMeter®

The QMeter® is a set of tools developed for call quality and bandwidth quality measurement.

3.1 Call quality

The call quality meter ensures that the network is meeting certain quality parameters. The basic parameter is signal strength, which has been measured for every five milliseconds of an active call and logged, to reveal if there is a change in the signal strength information. The signal strength classification is based on criteria in the table below. The average signal strength is calculated at the end of the call.

Signal level range (dBm)	**Classification**	**Score**
-120 to -95	Extremely bad	1
-95.00 to -85.00	Bad	2
-85.00 to -75.00	Average	3
-75.00 to -65.00	Good	4
-65.00 to -55.00	Very good	5

Table 1: Signal strength classification

The calls are classified as successful and unsuccessful call attempts, based on whether the call is successfully connected to the network. The successful attempts are again classified as normally dropped and dropped due to hand-over, which are the calls dropped during the cell change.

The call statistics for a bundle of 10 calls are considered for all the parameters. The successful call rate score is calculated based on the number of successful call attempts made for every 10 calls.

Successful call attempts	**Score**
1-2	1
7-8	2
5-6	3
3-4	4
1-2	5

Table 2: Successful call attempts score

The normal dropped rate score is classified based on the scale in Table 3.

Normally dropped rate	**Score**
< 4	1
>4 & <6	2
>6 & <7	3
>7 & <8	4
>8	5

Table 3: Normal dropped rate score

The average signal strength for all the successful calls is calculated together with successful call attempts score and normal dropped rate score for the bundle of 10 calls. The call quality is derived from the scores computed as:

(Average signal strength score of all successful calls + successful call rate score + normal dropped calls rate score)/3.

The final call quality for the bundle of 10 calls is classified according to the scale in Table 4.

Score	Classification
<1	Extremely bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

Table 4: Call quality score

The visualization of call quality is equally important from the perspective of both the operator and the end-user. The operator would be able to analyze the information from the end-user perspective and the user group can use it for deciding which operator to choose which can meet its requirements. As a part of the call quality meter, the average signal strength measured, with the score of the individual call, is landmarked on a map. The landmarks are marked in red if the calls are dropped due to hand-over. The landmarks are marked in green if the calls are normally dropped. The different colour landmarks help people to visualize and analyze calls.

Call quality escalation is another area of research which needs time to settle and to consolidate processes and procedures. A call quality meter has a built-in module, where the call quality escalation has been addressed from the end-user perspective, as an SMS to be sent to the mobile station integrated services digital network (MSISDN), which can be saved by the user. Mobile operators can promote this feature to encourage subscribers to participate. The module sends the average signal strength and score information to the particular number that has been saved for escalation. Call quality with respect to the average signal strength can be used either for a single call or for a bundle of 10. It can also support the setting of different options of when to escalate, such as escalate always, less than bad etc. Figure 1 illustrates the complete process of the signal meter system.

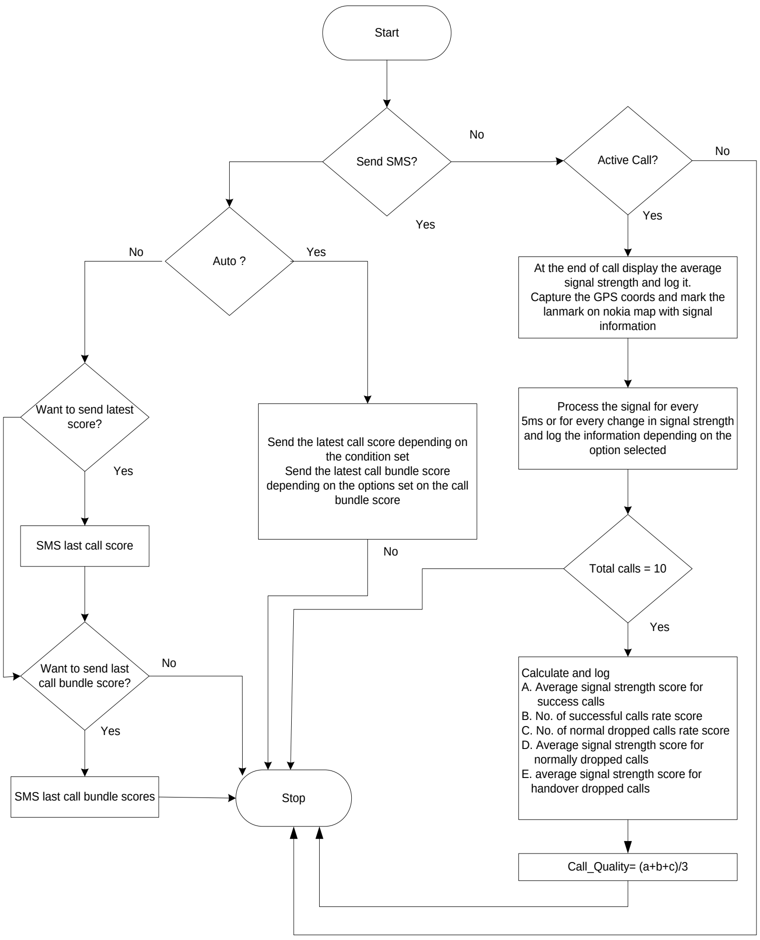

Figure 1: QMeter® call quality measurement flow chart

3.2 Bandwidth quality

The internet bandwidth quality provided by mobile operators fluctuates drastically and frustrates the user sometimes. As the number of users increases in a particular cell of the mobile network, the bandwidth decreases and hence there would be a loss of revenue, if the situation continued, for the operator. An attempt to measure the average bandwidth quality per individual and for bundles of 10 downloads is calculated as portrayed in Figure 2.

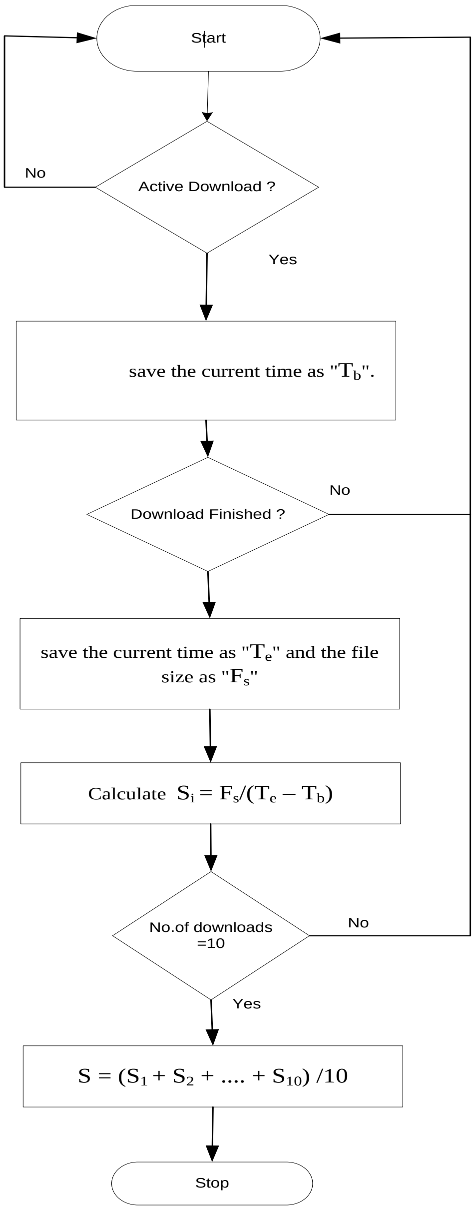

Figure 2: QMeter® bandwidth calculation flow-chart

The scores computed can be used by the user as well as the operator to evaluate bandwidth quality.

The average bandwidth of data download is computed using the following procedure:

- When a new download is initiated save the current time as "T_b".
- When the download is finished save the current time as "T_e" and the file size as "Fs".
- Calculate the average speed for this download as $S_i = F_s/(T_e – T_b)$.
- The average bandwidth for 10 downloads is calculated as $S = (S_1 + S_2 + ... + S_{10}) /10$.

The score for the average bandwidth is computed on the scale of 1-5.

Bandwidth	Score
< 32 kbps	1 (Extremely bad)
<32 & <64 kbps	2 (Bad)
<64 & < 128	3 (Average)
<128 & <256	4 (Good)
> 256	5 (Excellent)

Table 5: Bandwidth score

The bandwidth quality scores will give the user and the operator better insight into the usage of bandwidth. The approach can further be enhanced by capturing the cell-id and sending the critical scores for analysis to provide a better service.

4. Correlating with subjective scores

The results from the call quality meter are compared with the MOS (mean opinion scores) of the same calls for which the call quality scores are computed using the SM (signal meter). For each individual call the MOS is observed and classified based on Table 6.

MOS	Quality
1	Extremely bad
2	Bad
3	Average
4	Good
5	Excellent

Table 6: MOS classification

The classifications for MOS and SM are similar. Hence the average call quality computed for the calls mentioned below is compared with subjective average scores. The comparison is done in two stages, as shown in Table 7 and Table 8.. This is to ensure that the call quality scores correlate with MOS scores in all cases, from a low number of calls to a high number of calls at different locations.

No. of calls	MOS (average) (X)	Rank for X	MOS quality	QMeter average call quality (Y)	Rank for Y	QMeter quality
10	3	1	Average	2.7	1	Average
20	4	2.5	Good	3.8	2	Good
30	4	2.5	Good	3.9	3	Good
40	5	4.5	Excellent	4.8	4.5	Excellent
50	5	4.5	Excellent	4.8	4.5	Excellent

Table 7: Call quality versus MOS

The Spearman rank correlation for X (MOS scores) and Y (SM scores) computed for n=5 using the formulae below evaluates to 0.9733. The r_s= 0.9733 can be interpreted as MOS scores and SM scores are highly correlated with each other.

$$\rho = \frac{\sum_i (x_i - x)(y_i - y)}{\sqrt{\sum_i (x_i - x)^2 \sum_i (y_i - y)^2}}$$

No. of calls	MOS (average) (X)	Rank for X	MOS quality	QMeter average call quality (Y)	Rank for Y	QMeter quality
100	4	1	Good	3.6	1	Good
200	5	3.5	Excellent	4.6	2	Excellent
300	5	3.5	Excellent	4.8	3.5	Excellent
400	5	3.5	Excellent	4.9	5	Excellent
500	5	3.5	Excellent	4.8	3.5	Excellent

Table 8: Call quality versus MOS

The correlation coefficient is computed for n=5, the number of calls in the multiple of 100 evaluated to 0.7255. The r_s= 0.7255 can be interpreted as showing that MOS scores and SM scores are highly correlated with each other, but the correlation has slid to some extent due to an increase in the number of calls.

The correlation between SM call quality and MOS scores shows that SM quality scores are very close to the MOS listening scores. Therefore, the SM can be used to carry out subjective evaluation of call quality instead of using a human being, which is tedious and requires lot of time.

5. **Quality versus tariff**

The proposed tariff structures as per the parameters are improved in the version proposed in [13] and for the final call quality are computed in a bundle of 10 calls. The variable X is the normal charging rate per minute, n is the number called minutes in the bundle of 10 call attempts.

Call quality	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 9: Proposed charging rate versus call quality

6. **Conclusion**

The QMeter® research is aimed at deriving call quality and bandwidth quality parameters from the end-user's perspective in a mobile telecommunication network. The call quality parameters proposed use the signal strength and call drop information to measure the overall call quality. Call quality visualization techniques and escalation procedure are also proposed, which help mobile operators and user-groups to address quality issues. The correlation between subjective and QMeter® scores emphasizes the reliable nature of the QMeter®. The QMeter® can also be used by telecom regulatory authorities to monitor mobile operators' licence criteria for the quality of a network from the end-user's perspective. Further, it can be used as consumer protection tool to ensure that tariffs correlate with call quality.

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From the *Journal in Selected Areas of Telecommunication* (JSAT), May 2011.

Call Quality Measurement and Application in Telecommunication Network

Akram Aburas and Professor Khalid Al-Mashouq

1. Introduction

Traditional speech quality measurement techniques use a subjective listening test called mean opinion score (MOS). It is based on human perceived speech quality rated on a scale of 1 to 5, where 1 is the lowest perceived quality and 5 is the highest.

Subjective listening tests are expensive, time consuming and tedious. Currently, therefore, most systems use objective evaluation of speech quality using mobile computing techniques. Objective testing systems use automated speech quality measurement techniques. Three well known objective measurement techniques are perceptual speech quality measure (PSQM), the perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ).

2. Related work

Objective speech quality measurement techniques are mostly based on the input-output approach [1]. In input-output, objective measurement techniques basically work by measuring the distortion between the input and the output signal. The input signal would be a reference signal and output signal a received signal.

Input-output based speech quality assessment in objective speech quality measurement gave good correlations, with reaches up to 99 per cent in some cases [2]. Estimating speech quality without the presence of the input or reference signal is the latest area of research.

Jin Liang and R. Kubichek [3] published a paper on output-based objective speech quality using perceptually-based parameters as features. The results showed 90 per cent correlation. R. Kubichek and Chiyi Jin [4] used the vector quantization method, which showed 83 per cent correlation.

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A time-delay multilayer neural network model for measuring output based speech quality was proposed by Khalid Al-Mashouq and Mohammed Al-Shaye in [7]. The correlation achieved for speaker and text independently was 0.87.

In this paper we present our work for determining call quality parameters such as average signal strength, successful call rate and successful hand-over rate with respect to signal strength and successful call rate. Final call quality is computed from the extracted parameters.

This research is a continuation of the work that has been written up in [8-14]. A basic bandwidth quality measurement is proposed that can be used by both the operator and the user to evaluate the bandwidth quality of a particular operator.

3. Call quality

The research is focused on call quality measurement. Measuring call quality to ensure the quality of a mobile network and its reliability is essential. The proposed system is the outcome of our rigorous research to ensure that the network is meeting certain quality parameters.

The system logs the signal strength information for every five milliseconds if there is change in the signal strength information. The system records the number of successful and unsuccessful call attempts made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. Call drop information, such as calls normally dropped from either party or dropped due to hand-over during the cell change, is also recorded.

The average signal strength of successful calls, normal dropped and hand-over dropped calls, with their average scores, is recorded. The overall successful call rate score is derived based on the scale below:

Successful calls 9-10 score : 5 (excellent)

Successful calls 7-8 score : 4

Successful calls 5-6 score : 3

Successful calls 3-4 score : 2

Successful calls 1-2 score : 1 (very bad)

The normally dropped call rate score is derived based on the scale below:

Normal dropped calls >8 score : 5 (excellent)

Normal dropped calls < 7 & <8 score : 4

Normal dropped calls < 6 & < 7 score : 3

Normal dropped calls < 4 & < 6 score : 2

Normal dropped calls < 4 score : 1 (very bad)

Call quality, derived from the scores above, is computed as below:

(Average signal strength score of all successful calls + successful call rate score + normal dropped calls rate score)/3 .

The landmarks marked in red are the calls dropped due to hand-over and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls.

The system has the ability to send signal strength information to a particular mobile number. It can set the mobile number to which an SMS would be sent automatically at the end of the call. The system can send the SMS always, at the end of 10 calls or at other intervals. Figure 1 illustrates the complete process of the signal meter system.

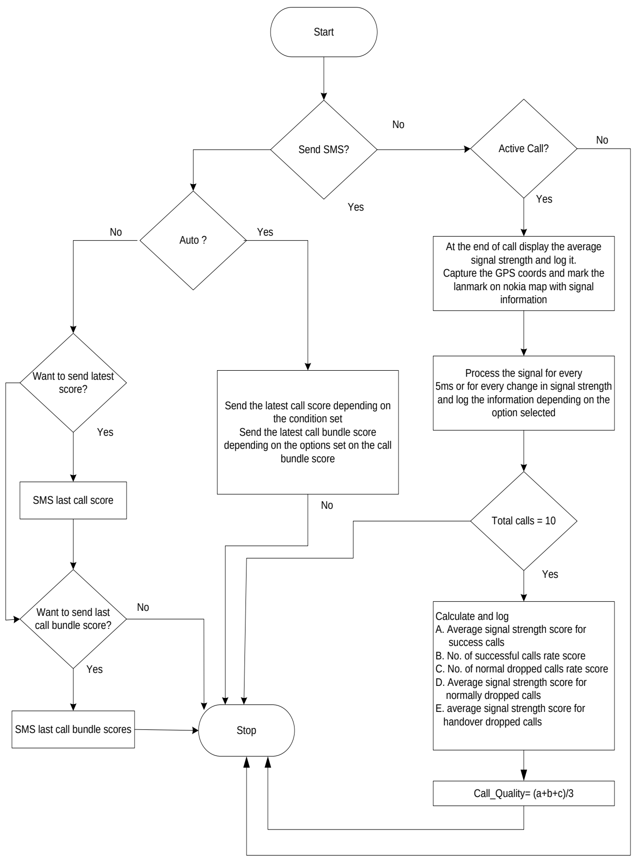

Figure 1: Signal meter flow chart

Signal_Measure( )

A generic algorithm of our signal measure algorithm is presented below:

1. Get the preferences for log_change, log_location

2. Get total_calls, Call_attempts_failed, call_attempts_successful, normal_dropped_calls, hand-over_dropped calls

3. if (total_calls =10) reset all variables to zeros

4. if (call_attempt = failed)

total_calls=total_calls+1

call_attempts_failed=call_attempts_failed+1

5. While (phone_status != idle && call_attempt = successful)

6. total_calls=total_calls+1

7. call_attempts_successful=call_attempts_successful+1

8. if(gps_coords available)

Get the gps_coords

9. Get the date, time, cell_id.

10. Get the signal_strength

11. if (log_change= 5ms)

write("signal_measure.log",date, time, gps_coords, cell_id, signal_strength)

12. if (log_change= when changed) and (signalstrengthchange=yes)

Write("signal_measure.log",date, time, phone status, signal strength)

13. End of While

14. End of call

15. Calculate average_signal_strength

16. If (average_signal_strength <= -95 && average_signal_strength >= -120) SignalQuality= Extremely Bad

Elseif (average_signal_strength <= -85 && average_signal_strength >-95) SignalQuality=Bad

Elseif (average_signal_strength <= -75 && average_signal_strength > -85) SignalQuality=Average

Elseif (average_signal_strength <=-65 && average_signal_strength > -75) SignalQuality=Good

Elseif (average_signal_strength <= -55 && average_signal_strength > -65)
SignalQuality=Very Good

17. Write("signal_measure.log",date, time, phone status, average_ signal_strength)

18. if (sendSMS = auto && whenSMSsend = always || sendSMS =auto && whenSMSsend < bad)

sendSMS(average_signal_strength,

SignalQuality,call_drop_information)

19. Write (SignalQuality)

20. If (GPS_Coords Available)

If (call_dropped = Normal)

Normal_dropped_calls=Normal_dropped_calls+1

Landmark_colour = green

Else

landmark_colour = red

hand-over_dropped_calls= hand-over_dropped_calls+1

Open(nokia_map)

Plot(gps_coords, landmark)

21. if (total_calls = 10)

Score_hand-over_dropped=

sum(hand-over_dropped_quality)/hand-over_dropped_calls

Score_normal_dropped=

sum(normal_dropped_quality)/normal_dropped_calls

score_successful_attempts=

(sum(hand-over_dropped_quality+sum(normal_dropped_quality))
/total_successful_attempts

22. If (call_attempts_successful< =2) Score_successful_call_rate = 1

Elseif (call_attempts_successful< =3 && average_signal_strength >=4)
Score_successful_call_rate = 2

Elseif (call_attempts_successful< =5 && average_signal_strength >=6)
Score_successful_call_rate =3

Elseif (call_attempts_successful< =7 && average_signal_strength >=8)
Score_successful_call_rate =4

Elseif (_attempts_successful< =9 && average_signal_strength >=10) Score_successful_call_rate =5

23. If (hand-over_success_calls< 40 per cent) Score_hand-over_success_calls_rate = 1

Elseif (hand-over_success_calls <40 per cent && hand-over_success_calls >60) Score_hand-over_success_calls_rate = 2

Elseif (hand-over_success_calls <60 per cent && hand-over_success_calls >70) Score_hand-over_success_calls_rate =3

Elseif (hand-over_success_calls <70 per cent && hand-over_success_calls >80) Score_hand-over_success_calls_rate =4

Elseif (hand-over_success_calls >80) Score_hand-over_success_calls_rate =5

24. Calculate average_call_quality=(score_successful_attempts+score_successful_call_rate+score_hand-over_success_calls_rate)/3

25. Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful, score_successful_attempts, normal_dropped_calls,score_normal_dropped, hand-over_dropped_calls,score_hand-over_dropped,score_successful_call_rate,score_hand-over_success_calls_rate, average_call_quality)

26. if (sendSMS = auto && whenSMSstat_send = always || sendSMS =auto && whenSMS_call_failed < 5) || whenSMS_hand-over_dropped < 2)

sendSMS(num_calls_unsuccessful,

num_calls_successful,

num_of_calls_dropped_hand-over,

num_normal_dropped)

27. If (log location = internal memory)

save signalmeter.log to c:/data

save calls_stats.log to c:/data

else save signalmeter.log to e:/data

save calls_stats.log to e:/data

28. if(sendSMS = Manual && want_to_send_SMS= yes)

set(mobile_number)

sendSMS(signal_stength, SignalQuality,call_drop_information)

29. End of Program

Table 1 shows final call quality classification based on the score for a bundle of 10 calls.

Score	Classification
<1	Extremely bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

Table 1: Call quality score

4. Correlating with subjective scores

The signal meter (SM) results are compared with the MOS (mean opinion scores) of the same calls for which the call quality scores are computed using SM. For each individual call the MOS is observed and classified based on Table 2. The classification for MOS and SM are similar. Hence the average call quality computed can be compared with subjective average scores. The comparison is done in two stages, as shown in Table, Table 3 and Table 4. This is to ensure that the call quality scores correlate with the MOS scores in all cases, from a low number of calls to a high number of calls, at different locations.

MOS	Quality
1	Extremely bad
2	Bad
3	Average
4	Good
5	Excellent

Table 1: MOS classification

No. of	MOS (average)	MOS quality	SM (average	SM quality

calls			**call quality)**	
10	3	Average	2.7	Average
20	4	Good	3.8	Good
30	4	Good	3.9	Good
40	5	Excellent	4.8	Excellent
50	5	Excellent	4.8	Excellent

Table 3: Call quality versus MOS

No. of calls	**MOS (average)**	**MOS quality**	**SM (average call quality)**	**SM quality**
100	4	Good	3.6	Good
200	5	Excellent	4.6	Excellent
300	5	Excellent	4.8	Excellent
400	5	Excellent	4.9	Excellent
500	5	Excellent	4.8	Excellent

Table 4: Call quality versus MOS

The correlation between SM call quality and MOS scores shows that SM quality scores are very close to the MOS listening scores. Therefore, the SM can used to carry out the subjective evaluation of call quality instead of using a human being.

5. Charging rate versus quality

Table 5 is the new charging rate proposed, based on the average signal strength of successful calls in a bundle of 10. The variable X is the normal charging rate per minute, n is the number of called minutes in a bundle of 10 call attempts.

Average signal strength of successful calls score	**Charge**
5	X*n
4	X*n
3	X*n*0.75
2	X*n*0.5
1	No charge

Table 5: Proposed charging rate versus average signal strength of successful calls

Table 6 is the new charging rate proposed, based on successful call attempts in a bundle of 10.

Successful call attempts score	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 6: Proposed charging rate versus successful call attempts

Table 7 is the new charging rate proposed, based on average signal strength of normal dropped calls in a bundle of 10.

Normal dropped rate score	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 7: Proposed charging rate versus average signal strength of normal dropped calls

Table 8 is the new charging rate proposed, based on total call quality of calls in a bundle of 10.

Call quality	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 8: Proposed charging rate versus call quality

6. Simulation results

2011/04/21 - 13:20:35 : : Current network info

LocationAreaCode = 352 CellId = 12211

2011/04/21 - 13:20:36 : : Signal strength is = 80 dBm, 7 bars

2011/04/21 - 13:20:58 : : Signal strength is = 83 dBm, 7 bars

2011/04/21 - 13:20:59 : : Signal strength is = 82 dBm, 7 bars

2011/04/21 - 13:21:07 : : Signal strength is = 77 dBm, 7 bars

2011/04/21 - 13:21:12 : : Signal strength is = 81 dBm, 7 bars

2011/04/21 - 13:21:44 : : Signal strength is = 79 dBm, 7 bars

2011/04/21 - 13:21:46 : : Signal strength is = 82 dBm, 7 bars

2011/04/21 - 13:21:47 : : Signal strength is = 78 dBm, 7 bars

2011/04/21 - 13:21:49 : : Call drop observer -> Event : Call state is changed. Phone status: Idle

2011/04/21 - 13:21:49 : : Average signal strength is 80 dBm (Average)

7. Sample call statistics

2011/04/24 - 07:45:33 : : 0 call attempts failed

2011/04/24 - 07:45:33 : : 10 call attempts successful :: Score: 3 (Average)

2011/04/24 - 07:45:33 : : 10 calls was normally dropped :: Score: 3 (Average)

2011/04/24 - 07:45:33 : : **********************

Figures 2 and 3 depict the landmarks of successful calls, with colours in green and red showing the normally and hand-over dropped calls.

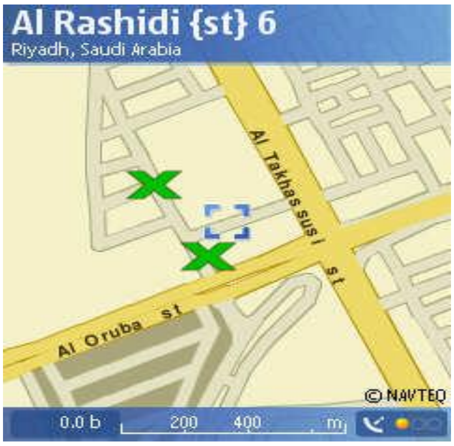

Figure 2: Landmarks for normally dropped calls

Figure 3: Landmarks for normally and hand-over dropped calls

8. Conclusion

The research presents a comprehensive amalgamation of research from different call quality measurement parameters, with final average call quality measurement, correlating the call quality scores with subjective scores, call quality escalation, landmarking the call quality, and tariff propositions based on call quality parameters. The research is useful for the telecom industry to understand call quality from the end-user's perspective and take the necessary measures to reduce customer "churn" and increase average revenue per client. The research could also be used by the telecom regulatory authorities to monitor whether operators are meeting required licence criteria of quality of network from end-user's perspective. Further, it can be used as a consumer protection tool to ensure that tariffs correlate with call quality.

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A Model for Call Quality Computation and Collection in Mobile Telecommunication Networks

Akram Aburas and Professor Khalid Al-Mashouq

1. Introduction

Traditional speech quality measurement techniques use a subjective listening test called mean opinion score (MOS). It is based on human perceived speech quality rated on a scale of 1 to 5, where 1 is the lowest perceived quality and 5 is the highest.

Subjective listening tests are expensive, time consuming and tedious. Currently, therefore, most systems use objective evaluation of speech quality using mobile computing techniques. Objective testing systems use automated speech quality measurement techniques. Three well known objective measurement techniques are perceptual speech quality measure (PSQM), the perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ).

Objective speech quality measurement techniques are mostly based on the input-output approach [1]. In input-output, objective measurement techniques basically work by measuring the distortion between the input and the output signal. The input signal would be a reference signal and output signal a received signal.

Input-output based speech quality assessment in objective speech quality measurement gave good correlations, with reaches up to 99 per cent in some cases [2]. Estimating the speech quality without the presence of the input or reference signal is the latest area of research.

Jin Liang and R. Kubichek [3] published a paper on output-based objective speech quality using perceptually-based parameters as features. The results showed 90 per cent correlation. R. Kubichek and Chiyi Jin [4] used the vector quantization method, which showed 83 per cent correlation.

An output based speech quality measurement technique using the visual effect of a spectrogram is proposed in [5]. An output-based speech quality evaluation algorithm based on characterizing the statistical properties of speech spectral density distribution in the temporal and perceptual domains is proposed in [6]. The correlations achieved with subjective quality scores were 0.897 and 0.824 for the training data and testing data set respectively.

A time-delay multilayer neural network model for measuring output based speech quality was proposed by Khalid Al-Mashouq and Mohammed Al-Shaye in [7]. The correlation achieved for speaker and text independently was 0.87.

In this paper we present our work for determining call quality parameters such as average signal strength, successful call rate and successful hand-over rate with respect to signal strength and successful call rate. Final call quality is computed from the extracted parameters.

This research is the continuation of the work that has been described in [8-16].

2. Call quality computation and escalation

The research parameters and the research method address call quality from the end-user's perspective. The system logs the signal strength every five milliseconds if there is a change in the signal strength, and calculates the score every five milliseconds. It also calculates the average signal strength at the end of the call. Call drop information – such as normally dropped from either party or dropped due to network issues or during cell change – is also recorded.

Signal range (dBm)	**Classification**
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Excellent

Table 1: Signal strength score

Call drop	**Score**
Drop due to network issue	1 (Extremely bad)
Normally dropped	5 (Excellent)

Table 2: Call drop score

Call quality is derived from the scores computed from these parameters as below:

$$\text{Call quality} = (\text{Signal strength score}+\text{Call drop score})/2$$

Table 3 shows the final call quality classification

Score	**Classification**
<1	Extremely Bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

Table 3: Call quality score

The basic flow chart for the call quality computation is depicted in Figure 1.

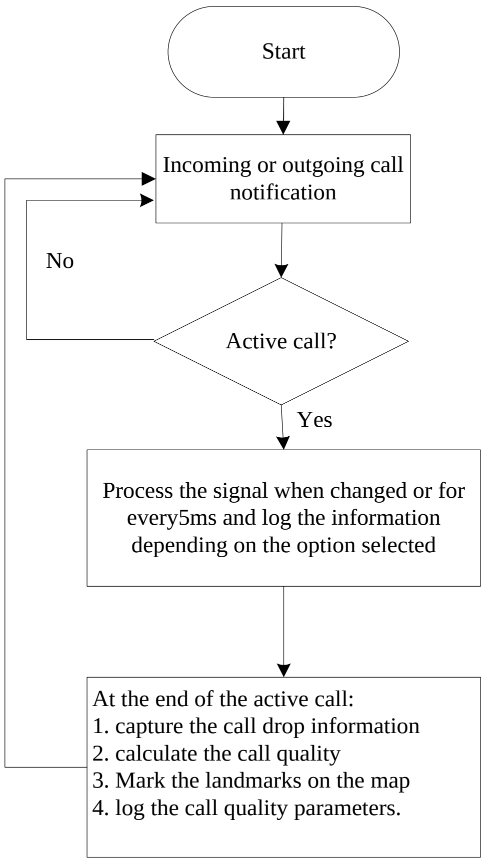

Figure 1: Call quality computation

The application uses GPS co-ordinates to mark landmarks. It also uses the LAC (location area code) and cell identification related to that particular operator. The LAC and cell IDs are more meaningful to the operator, if it wants to view call quality on maps at its end. The landmark for each call on the mobile equipment is marked on the map with colours. The landmarks that are marked in red are the calls dropped due to network issues and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze calls.

The system has the ability to send call quality information to a particular number as a call quality escalation. It has the provision of setting the mobile number to which the SMS would be sent automatically at the end of a call. The system has the option of setting to send the SMS always, when the call is less than bad, when it is bad and so on. It also has the provision to use call quality based on call statistics, where the call quality is computed at the end of 10 calls and the call statistics can be sent as SMS. This flexibility allows the operator to fix the parameter values that can be escalated for immediate action. The SMS call quality module is depicted in Figure 2.

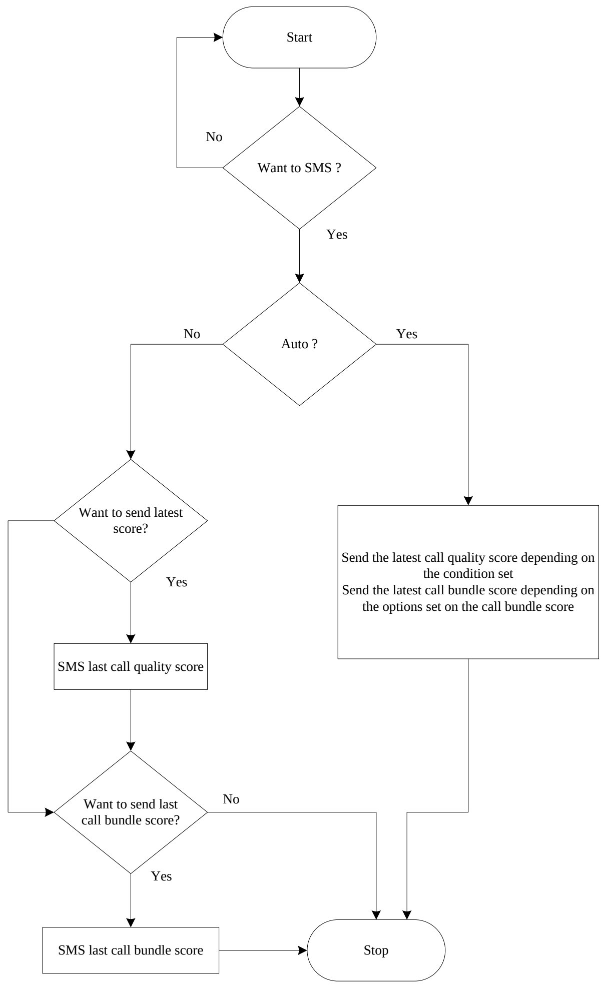

Figure 2: SMS call quality

3. Collection model for call quality

The average signal strength score and call drop score are calculated and the final call quality score is derived as per the average scores proposed in the previous section. At the end of the call, all the parameters, cell ID and GPS co-ordinates will be sent by SMS to some predefined short code at the operator end. All the information that has been collected at the operator end will be stored in a database. This will help the operator to analyze call quality for benchmarking and addressing other issues related to call quality. Various reports can be derived based on different parameters at different cell IDs and GPS co-ordinates, which is useful for operators. Further, call quality below threshold, as perceived by the end-user, can be used by the operator to apply tariff redemption or to add bonus amount or minutes to subscribers' accounts based on the policies defined by the service provider. A block diagram showing the flow of call quality from mobile equipment to QMeter® application at the operator is shown in Figure 3. Also the tariff redemption and bonus minutes can be used as a marketing tool by operators. Different tariff redemption methods are proposed in [14] based on call quality parameters.

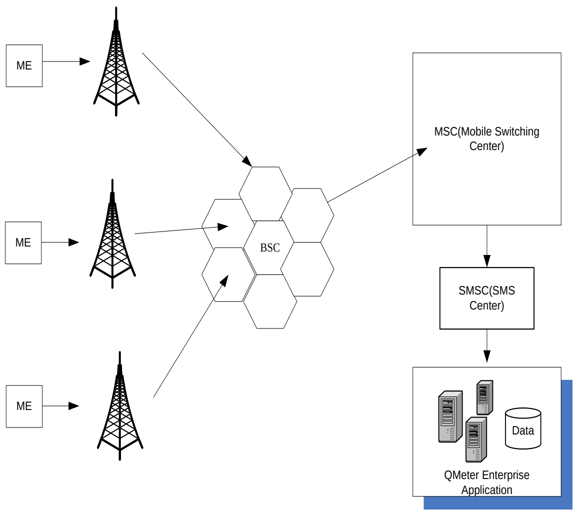

Figure 3: Call quality collection model

QMeter® is a set of tools for call quality measurement developed as an enterprise application, which collects call quality parameters from the end-user mobile equipment to analyze and process call quality.

Acknowledgment

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C2 Conference papers

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Call Quality versus Subjective Scores

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1. Introduction

Traditional speech quality measurement techniques use a subjective listening test called mean opinion score (MOS). It is based on human perceived speech quality, rated on a scale of 1 to 5, where 1 is the lowest perceived quality and 5 is the highest.

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Objective speech quality measurement techniques are mostly based on the input-output approach [1]. In input-output, objective measurement techniques basically work by measuring the distortion between the input and the output signal. The input signal would be a reference signal and output signal a received signal.

Input-output based speech quality assessment in objective speech quality measurement gave good correlations, with reaches up to 99 per cent in some cases [2]. Estimating the speech quality without the presence of the input or reference signal is the latest area of research.

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A time-delay multilayer neural network model for measuring output based speech quality was proposed by Khalid Al-Mashouq and Mohammed Al-Shaye in [7]. The correlation achieved for speaker and text independently was 0.87.

In this paper we present the results of our work on call quality measurement. This research is a continuation of the work that has been described in [8-14]. The work presented here is the final result of the call quality approach presented in [15]. A brief discussion of the work presented in [15] is included in Section 2.

2. Call quality

This paper is focused on call quality measurement scores proposed in [15] and their correlation with subjective scores. Call quality is extracted based on parameters discussed in [15]. The system proposed in [15] has been named signal meter (SM). The system proposed logs the signal strength information every five milliseconds, if there is change in the signal strength information. The system records the number of successful and unsuccessful call attempts made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The call drop information – such as normally dropped from either party or dropped due to hand-over during the cell change – is also recorded.

The average signal strength of successful calls, normal dropped and hand-over dropped, with their average scores, is recorded. The overall successful call rate score is also derived based on the scale below:

Successful calls 9-10 score : 5 (excellent)

Successful calls 7-8 score : 4

Successful calls 5-6 score : 3

Successful calls 3-4 score : 2

Successful calls 1-2 score : 1 (very bad)

The normally dropped call rate score is derived based on the scale below:

Normal dropped calls >8 score : 5 (excellent)

Normal dropped calls < 7 & <8 score : 4

Normal dropped calls < 6 & < 7 score : 3

Normal dropped calls < 4 & < 6 score : 2

Normal dropped calls < 4 score : 1 (very bad)

The average signal strength for all the successful calls is calculated together with successful call attempts score and normal dropped rate score for the bundle of 10 calls. The call quality is derived from the scores computed as:

(Average signal strength score of all successful calls + successful call rate score + normal dropped calls rate score)/3.

The final call quality for the bundle of 10 calls is classified according to the scale in Table 1.

Score	**Classification**
<1	Extremely bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

Table 1: Call quality score

3. Call quality versus MOS scores correlation

The signal meter (SM) results are compared with the MOS (mean opinion scores) of the same calls for which the call quality scores are computed using SM. For each individual call the MOS is observed and classified based on Table 2. The classifications for MOS and SM are similar. Hence the average call quality computed can be compared with subjective average scores. The comparison is done in two stages, as shown in Table 2 and Table 3. This is to ensure that the call quality scores correlate with the MOS scores in all cases, from a low number of calls to a high number of calls, at different locations.

MOS	**Quality**
1	Extremely bad
2	Bad
3	Average
4	Good
5	Excellent

Table 2: MOS classification

No. of calls	**MOS (average)**	**MOS quality**	**SM (average call quality)**	**SM quality**
10	3	Average	2.7	Average
20	4	Good	3.8	Good
30	4	Good	3.9	Good
40	5	Excellent	4.8	Excellent
50	5	Excellent	4.8	Excellent

Table 3: Call quality versus MOS

No. of calls	**MOS (average)**	**MOS quality**	**SM (average call quality)**	**SM quality**
100	4	Good	3.6	Good
200	5	Excellent	4.6	Excellent
300	5	Excellent	4.8	Excellent
400	5	Excellent	4.9	Excellent
500	5	Excellent	4.8	Excellent

Table 4: Call quality versus MOS

The correlation between SM call quality and MOS scores shows that SM quality scores are very close to the MOS listening scores. Therefore, the SM can used to carry out the subjective evaluation of call quality instead of using a human being.

4. Conclusions

The results of the research in [15] correlate with the subjective scores, which underlines that the approach presented could reliably used to simulate subjective listening tests and hence would ease the laborious process involving large numbers of people testing telecommunication networks. Further, the tool can be used by the operators, as mentioned in [15], and as a consumer protection tool to ensure that tariffs correlate with call quality.

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Real Time Speech Quality Analysis in a Wireless Network

K. Al-Mashouq, A. Aburas and M. Maqbool

1. Introduction

This paper focuses on arriving at an approach that uses transmission parameters on a GSM network to determine voice or speech quality on a communication network. Traditionally, theories have mainly focused on assessing speech quality in terms of mean opinion score (MOS) which was graded accordingly on a scale based on a human perceived level. This method is not practical to be used with a live network.

This contribution aims at deducing a technique which, through the use of GSM parameters, enables us to show that it is possible to correctly predict the speech quality score. Parameters such as signal level, bit error rate, frame erasure rate and carrier to interference (C/I) ratio have been used in extensive simulations to quantify the speech quality score.

2. Speech quality assessment method

This paper focuses on obtaining speech quality scores through the use of GSM parameters that are transmitted during an active call on a GSM network. In this endeavour, it has been attempted to secure a correlation between the parameters that are being transmitted and the perceptual evaluation of speech quality (PESQ) [1] which is further used to show that these parameters can help to arrive at a speech quality score without the use of the original speech sample or the received speech sample. It is further been demonstrated that the correlation based on these parameters is very close.

Speech samples that are transmitted during an active conversation were obtained and associated parameters with each sample – such as PESQ score, signal level, bit error rate, frame erasure rate and C/I – were also obtained from the same samples. The speech samples obtained from an active conversation were given to listeners with normal hearing. Listeners were trained to familiarize themselves with the different versions of samples received, either in excellent or distorted form. Scoring of speech samples by listeners was undertaken in which, after hearing each speech sample, the listeners had to grade the speech sample they had listened to.

Score	**Classification**
<1	Extremely bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

Table 1: Grading score of speech samples

Table 1 outlines the grading, or PESQ score, of the speech samples and the associated parameters mentioned earlier. Listeners, after having been trained on some speech samples, were asked to classify the samples as per the table above.

Every speech sample obtained or used in this exercise constitutes a five second voice transmission. It is to be noted that PESQ is basically a reference based modeVmethod, that uses the received signal alone, extracts parameters from it and tries to assess to what degree the distortions in the received signal will be audible to the human ear. The approach of this paper is likewise aimed at attempting to correlate and successfully link the correlation between the obtained parameters and the speech samples graded by the listeners [2-4].

As shown further, we aim at statistically correlating and successfully linking the correlation between the obtained signal parameters and the speech samples graded by the listeners.

Rx.level	Rx.Qual	FER	*Cil*	PESO
-86.56	0.71428	1.28	20	3.9
83.3 I	0	1.5	20	3.9
78.33	0	I	20	3.9
8I.97	0	1.333	20	3.9
82.76	0.57142	0.9411	20	3.9
-83.10	0	1.2	20	3.9

Table 2: Sample of data obtained from test equipment

It is perhaps worth mentioning that the reason for having human listeners grade the speech sample, in addition to the measuring system, is the necessity to see the difference between a system grading and a human ear grading and then to develop a model which estimates speech scores as perceived by the human ear.

Rx. Qual is one parameter that is basically a mapping of time averaged bit errors over a scale of 0 to 7, which gives a rough indication of speech quality. During the analysis, it was found that, for every speech sample of five seconds' duration, numerous

Rx. Qual values were obtained, which is quite logical given the duration of the transmission and the fact that the measuring system undertakes the measurement many times. The Rx. Qual values (or the bit error rate values) were averaged out to gain an average for the whole transmitted sample.

Frame erasure rate is a parameter that is critical in speech quality assessment as it is a measure of the percentage of erased frames compared to number of frames transmitted. As with the Rx. Qual, we obtained the frame erasure rate for every sample and the values were also averaged out here.

Carrier to interference ratio helps in determining the level of interference the subjected signal has undergone. A high C/I will indicate a good signal and yield a good communication signal quality, whereas a low C/I will result in the opposite.

Keeping in view the above parameters and their significance, extensive simulations were carried out on all the samples obtained to give a correct estimate of speech quality.

We used a liner model to estimate the PSEQ score using the following four parameters: RxLevel (or signal strength), Rx.Qual, C/I and FER. They are combined using least square method. This model is then tested with real data and compared with PSEQ.

The estimated quality score, q, will be:

q=2:aiwi where; i ranges from 1 to 4 Wi is the weighing factor al is Rx. Level a~ isRx. Qual a3isFER <4 is C/I

The standard least square solution to this problem is given by:

~=(ATArIAT~
where; ~ = (WI w~ W3 W4)T

[
all al~a13aI
J
A- aLl a,~3;3a,4
aNI aN~ aN> aN4

Here, (ail ai2 3;3 ai4) corresponds to the measurement ofRx. Level, Rx. Qual, FER and C/I respectively. For, the vector .G.-= (C1 C2 ""... CN)T; Ci corresponds to the ith sample of PESQ measurement.

3. **Results**

Extensive simulations on the obtained speech samples were performed. The correlation between the estimated quality score and PESQ value obtained from the measurement system ranges between 90 per cent and 95 per cent.

4. **Conclusion**

The results of this research, which aims to correctly estimate the speech quality of signals without making use of the transmitted sample or the original sample, reveal an accuracy of more than 90 per cent with the above mentioned parameters. This paper has further outlined the possibility of making speech quality analysis through the use of transmitted GSM parameters and gives a result that is as perceived by the human ear.

It is our understanding that this contribution could provide a cost effective method of evaluating speech quality by using only the parameters that are transmitted on a GSM.

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Perceptual Evaluation of Call Quality and Evaluation of Telecom Networks

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1. Introduction

Traditionally, users' perception of speech quality is measured offline using subjective listening tests. The method of subjective testing called mean opinion score (MOS) provides a numerical indication of the perceived quality of received human speech over the connection. The MOS is expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality and 5 is the highest. But subjective estimation by repeated listening tests at various sites within the coverage area is laborious, expensive and time-consuming. It would be much more desirable to use an automatic objective evaluation system that applies a good objective speech quality measure to estimate the statistical average of subjective opinions of the typical conversational speech sentences sent through the mobile network.

Objective assessment of the speech quality in modern communication systems is typically achieved by measuring some form of distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signal power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

The results of MOS testing are expensive and impractical for testing in the field. Automatic testing algorithms were created in an attempt to formulate objective network testing similar to signal to noise ratio (SNR), bit error rate (BER) and received signal strength (RSSI) which are used to measure the signal quality. Objective testing algorithms are also called automated quality measurement techniques. Three objective tests were developed, namely:

1. Perceptual speech quality measure (PSQM)
2. Perceptual analysis measurement system (PAMS)
3. Perceptual evaluation of speech quality (PESQ).

Perception models for evaluating speech quality were developed by Mike Hollier, at BT Labs. John Beerends, of KPN Research, led subsequent innovations in the 1990s on the use of perception for voice quality assessment [1, 2]. Hollier observed that taking account not just of the amount, but also the distribution, of audible distortion could make quality predictions much more accurate. His work was taken up in 1996 by Antony Rix and forms the core of PAMS [3]. It was not until 1996, following a lengthy international study, that those perceptual models for quality assessment were first standardized. The result of this was that Beerends' model, PSQM, became an ITU-T recommendation (P. 861) for assessing speech codecs [4].

Over the last decade, researchers and engineers in the field of objective measures of speech quality have developed different techniques based on various speech analysis models. Currently, the most popular techniques are those based on psychoacoustics models, referred to as perceptual domain measures [5]. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach [6]. In input-output objective assessment methods, the speech quality is estimated by measuring the distortion between an “input”, or a reference signal, and an “output”, or received signal. Using a regression technique, the distortion values are then mapped into estimated quality.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), modified Berkeley software distribution (MBSD), measuring normalizing blocks (MNB), the PSQM+, the telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS), and most recently the perceptual evaluation of speech quality (PESQ) [6], which is specified by ITU-T recommendation P. 862 [7], as the international standard for testing networks and codecs. Correlation between the objective speech quality measure and the subjective quality measure is mostly used as the system (or method) performance measure. In the case of input-output based speech quality assessment, good correlations were observed, reaching up to 99 per cent in some cases [8].

The field of estimating speech quality using only received speech without access to the input record is a relatively new area. Most recently, a couple of attempts to develop more credible non-intrusive speech quality measurements based on perceptual analysis have been reported. In 1994, Jin Liang and R. Kubichek [9] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results, in certain cases achieving 90 per cent correlation. Perceptually-based methods imitate the human listening method where many parameters and environmental effects are considered. R. Kubichek and Chiyi Jin [10, 11] used the vector quantization method, which yields up to 83 per cent correlation. Vector quantization has some disadvantages that yield poor results. One of these disadvantages is an inherent spectral distortion in representing the actual analysis vector. Since there is only a finite number of code book vectors, the process of choosing the “best” representation of a given spectral vector inherently is equivalent to quantizing the vector. It leads, by definition, to a certain level of quantization error. As the size of the code book increases, the quantization error decreases. However, with any finite code book there will always be some non-zero level of quantization error. Furthermore, the storage required for code book vectors is often significant. The larger we make the code book, the more storage space is required for the code book entries.

Another example of this is an output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal, reported in [12]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al.* [13] reported a novel use of vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in a non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [14]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

In [15] a new output-based speech quality measure, which uses bark spectrum analysis and fifth order perceptual linear prediction (PLP), was introduced. The measure is based on comparing the output speech to an artificial reference signal that is appropriately selected from the optimally clustered reference code book, using the self-organizing map approach coupled with an enhanced k-means technique. The average correlation of this technique reached 0.85 for bark spectrum and 0.61 for PLP coefficient, respectively. Both short duration of speech records and limited number of speakers count as a disadvantage of this study. Only two male speakers are used, one for training and the other one for testing.

In [16], Khalid Al-Mashouq and Mohammed Al-Shaye proposed a time-delay multilayer neural network model that can rate the speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPC) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached 0.87. However, the published result is not generalized due to the limited number of speech files used.

The latest research was published in 2004 [17]. This paper describes a new output-based approach that extends the original pseudo-reference framework by replacing the VQ code book with a discrete hidden Markov model. Correlation between objective and subjective scores shows good performance for text dependent and text independent cases, of 0.88 and 0.92 respectively, but does not provide speaker independence. However, none of these output-based researches are in the mobile communication environment except in the Al-Mashouq and Al-Shaye work [18].

In this work we propose a new scheme for measuring speech quality over the network from the subscriber set. This method adopts a non-traditional approach and can be used as a relative index for assessing the quality of speech. Further, the method offers a significant advantage over traditional speech quality measurement as the network operator and subscriber can evaluate the speech quality for every call made under different conditions. This can be applied in a number of ways:

1. In present day multi-operator environments, service subscribers can easily use

this method to choose between different networks.
2. Telecom regulatory bodies can apply this method to enact laws that will make operators charge less for less voice quality measured from their networks.
3. Network operators can use this as a marketing tool and offer subscribers reduced tariffs whenever the speech quality of calls they make is lower than a particular threshold.

The proposed method is part of ongoing research that will involve investigation of metrics which are used to detect the call, obtain the signal strength information, refresh the value at every five milliseconds and finally implement the application in a Symbian operating system.

2. The research method

Our ultimate goal in this work is to develop mobile handset software which will perform an automatic speech quality assessment for every telephone call. It should give an objective score for each call along with a periodic speech quality average. This, in turn, will give subscribers a practical way to assess the performance of different operators. Moreover, operators may want to give their employees a version of this software to do internal network auditing.

To achieve our objective, a continuous hidden Markov model (HMM)-based speech recognition system is used to generate phonetic segmentations of the processed speech. Based on these segmentations and statistical models, different probabilistic scores are derived for the processed speech to model human perception of the quality of communication channels. HMM is considered a basic component in speech recognition systems. The estimation of good model parameters affects the performance of the recognition process so the values of these parameters need to be estimated such that the recognition error is minimized. The model parameters are usually determined during an iterative process called the training process.

One of the conventional methods applied in setting HMM model parameters values is the Baum-Welch algorithm. This is one of the traditional iterative techniques that are used to estimate HMM parameters. The algorithm starts from an initial model and iteratively improves on it (updates) until convergence is reached. The algorithm is guaranteed to converge to an HMM that locally maximizes the likelihood (the probability of the training data given the likelihood).

As a starting point, we obtained a cellular speech database in its original version. Then, we built up a robust continuous speech recognizer. Finally, we derived automatic machine scores wherein the correlation with the input-output counterpart is very high. The speech databases used in most of the simulation stage are the well-known TIMIT and CTIMIT database set. CTIMIT is the cellular version of the TIMIT phonetic database. We choose these databases because of their large vocabulary and mixed speaker accents. The HMMs are trained using a clean speech database. The noisy speech database is used for testing. Each word of the database is segmented and phonemes assigned. Then that word is trained for the HMM model parameters using the Baum-Welch algorithm.

For test word recognition, we used the degraded speech samples and the forward or backward algorithm of HMM to find the most likely word in the database. The recognized word's probability was categorized on 1-5 scale of quality. Then the test scores were correlated with the PESQ algorithm.

This is a non-traditional approach. The aim is to use it as a relative index when assessing the quality of speech. As a step towards realization of our research, the software has been developed on a Symbian platform, which has been started as the signal strength measurement. This is applied in the following manner:

1. Upon a call being set up the signal level values are taken every five milliseconds.
2. The criteria in Table 1 are used to decide on the quality of the signal.
3. A score is then given to the sample collected every five milliseconds, with five being the best on a scale of 1 to 5 and 1 indicating very bad signal strength.
4. At the end of the call session a cumulative score is computed and based on the score (ranging from 1 to 5) the speech quality is approximately computed.

Signal level range (dBm)	**Classification**
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Signal quality

The signal strength parameters – such as number of normal dropped calls, dropped due to hand-over (cell hand-over), number of failed call attempts (due to network failure), successful call attempts – were also analyzed for a call bundle of 10. The call quality information was plotted as landmarks to visualize the information. The graphs below show the statistical information captured for two different networks based on our call quality parameters.

3. Emerging results

The emerging results of the research are shown in the graphs below. The networks of Operator A and Operator B were compared on our mentioned parameters. The best scores of both operators were the same, as shown in Figure 1. The worst scores of Operator A and Operator B are shown in Figure 2 and Figure 3. The call quality is measured based on a call bundle of 10. The experiment was carried out on 100 calls for each network over a period of six months. The call quality of the bundles was recorded by the software in the log file and the information was plotted as landmarks on the map by capturing the GPS co-ordinates of the location during the active call.

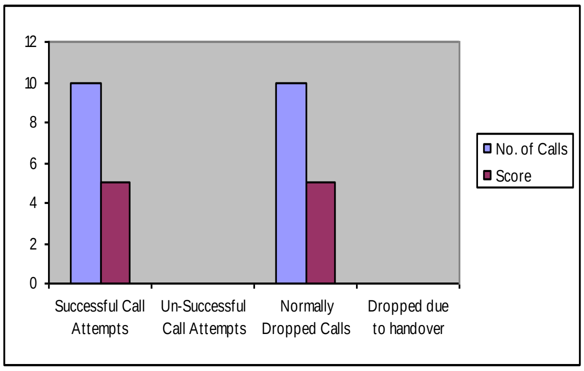

Figure 1: Operator A and Operator B best scores

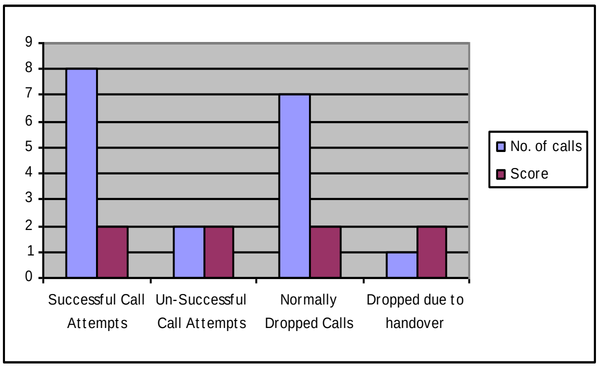

Figure 2: Operator A worst score

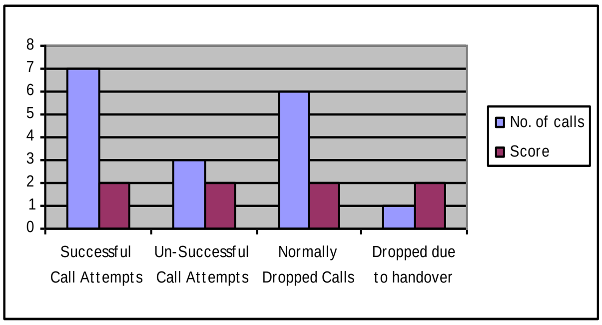

Figure 3: Operator B worst score

4. Summary

This report has examined a more liberal approach to speech quality measurement in telecommunication networks. It aims to equip network subscribers with the opportunity to choose their telecom service provider based, among other key indices, upon speech quality. This will also afford service providers the capability of predicting customers' opinion of quality of service and the need for necessary network optimization for continued customer satisfaction that will ensure loyalty and an increased subscriber base.

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Call Quality and its Parameters Measurement in Telecommunication Network

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1. Introduction

Traditional speech quality measurement techniques use a subjective listening test called mean opinion score (MOS). It is based on human perceived speech quality rated on a scale of 1 to 5, where 1 is the lowest perceived quality and 5 is the highest.

Subjective listening tests are expensive, time consuming and tedious. Currently, therefore, most systems use an objective evaluation of speech quality using mobile computing techniques. Objective testing systems use automated speech quality measurement techniques. Three well known objective measurement techniques are the perceptual speech quality measure (PSQM), the perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ).

Objective speech quality measurement techniques are mostly based on the input-output approach [1]. In input-output, objective measurement techniques basically work by measuring the distortion between the input and the output signal. The input signal would be a reference signal and output signal would be a received signal.

Input-output based speech quality assessment in objective speech quality measurement gave good correlations, with reaches up to 99 per cent in some cases [2]. Estimating the speech quality without the presence of the input or reference signal is the latest area of research.

Jin Liang and R. Kubichek [3] published a paper on output-based objective speech quality using perceptually-based parameters as features. The results showed 90 per cent correlation. R. Kubichek and Chiyi Jin [4] used the vector quantization method, which showed 83 per cent correlation.

An output based speech quality measurement technique using the visual effect of a spectrogram is proposed in [5]. An output-based speech quality evaluation algorithm based on characterizing the statistical properties of speech spectral density distribution in the temporal and perceptual domains is proposed in [6]. The correlations achieved with subjective quality scores were 0.897 and 0.824 for the training data and testing data set respectively.

A time-delay multilayer neural network model for measuring output based speech quality was proposed by Khalid Al-Mashouq and Mohammed Al-Shaye in [7]. The correlation achieved for speaker and text independently was 0.87.

In this paper we present our work for determining call quality parameters such as average signal strength, successful call rate and successful hand-over rate with respect to signal strength and successful call rate. Final call quality is computed from the extracted parameters.

This research is a continuation of the work that has been described in [8-14]. A basic bandwidth quality measurement is proposed which can be used by both the operator and the user to evaluate the bandwidth quality of a particular operator.

2. Call quality

The research is focused on call quality measurement. The system logs the signal strength information for every five milliseconds if there is change in the signal strength information. The system records the number of successful and unsuccessful call attempts made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The call drop information – such as normally dropped from either of the party or dropped due to hand-over during the cell change – is also recorded.

The average signal strength of successful calls, normal dropped and hand-over dropped calls, with their average scores, is recorded. The overall successful call rate score is also derived based on the scale below:

Successful calls 9-10 score : 5 (Excellent)

Successful calls 7-8 score : 4

Successful calls 5-6 score : 3

Successful calls 3-4 score : 2

Successful calls 1-2 score : 1(Very bad)

Normally dropped call rate score is derived based on the scale below:

Normal dropped calls >8 score : 5(Excellent)

Normal dropped calls < 7 & <8 score : 4

Normal dropped calls < 6 & < 7 score : 3

Normal dropped calls < 4 & < 6 score : 2

Normal dropped calls < 4 score : 1 (Very bad)

The call quality is derived from the scores computed from the above parameters, as below:

(Average signal strength score of all successful calls + successful call rate score + normal dropped calls rate score)/3 .

The landmarks marked in red are the calls dropped due to hand-over and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls.

The system has the ability to send signal strength information to a particular mobile number. It can set the mobile number to which an SMS would be sent automatically at the end of the call. The system can send the SMS always, at the end of 10 calls or at other intervals. Figure 1 illustrates the complete process of the signal meter system.

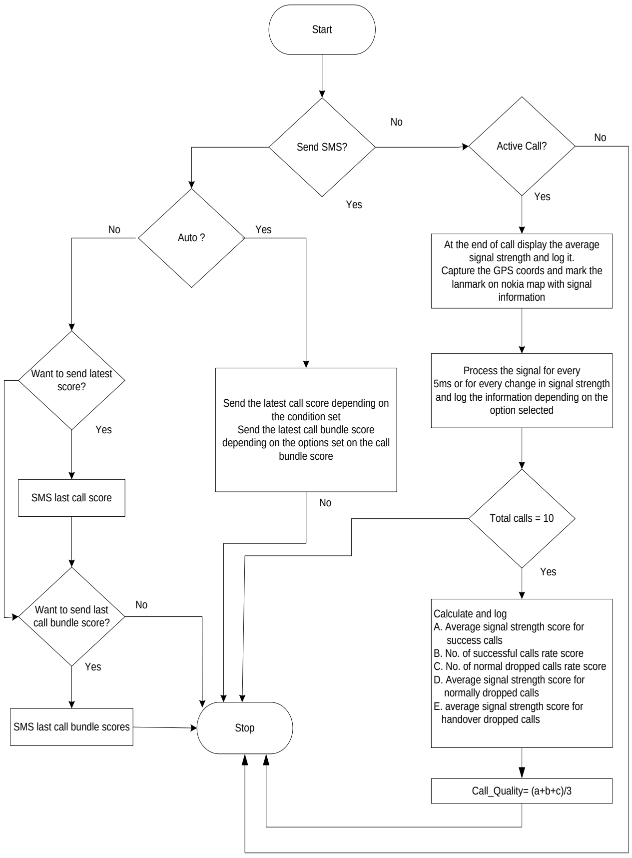

Figure 1: Signal meter flow chart

A generic algorithm of the signal measure algorithm is presented below:

Signal_Measure( )

1. Get the preferences for log_change, log_location

2. Get total_calls, Call_attempts_failed, call_attempts_successful, normal_dropped_calls, hand-over_dropped calls

3. if (total_calls =10) reset all variables to zeros

4. if (call_attempt = failed)

    total_calls=total_calls+1

call_attempts_failed=call_attempts_failed+1

5. While (phone_status != idle && call_attempt = successful)

6. total_calls=total_calls+1

7. call_attempts_successful=call_attempts_successful+1

8. if(gps_coords available)

    Get the gps_coords

9. Get the date, time, cell_id.

10. Get the signal_strength

11. if (log_change= 5ms)

write("signal_measure.log",date, time, gps_coords, cell_id, signal_strength)

12. if (log_change= when changed) and (signalstrengthchange=yes)

Write("signal_measure.log",date, time, phone status, signal strength)

13. End of While

14. End of call

15. Calculate average_signal_strength

16. If (average_signal_strength <= -95 && average_signal_strength >= -120) SignalQuality= Extremely Bad

Elseif (average_signal_strength <= -85 && average_signal_strength >-95) SignalQuality=Bad

Elseif (average_signal_strength <= -75 && average_signal_strength > -85) SignalQuality=Average

Elseif (average_signal_strength <=-65 && average_signal_strength > -75) SignalQuality=Good

Elseif (average_signal_strength <= -55 && average_signal_strength > -65) SignalQuality=Very Good

17. Write("signal_measure.log",date, time, phone status, average_ signal_strength)

18. if (sendSMS = auto && whenSMSsend = always || sendSMS =auto && whenSMSsend < bad)

sendSMS(average_signal_strength,

SignalQuality,call_drop_information)

19. Write (SignalQuality)

20. If (GPS_Coords Available)

If (call_dropped = Normal)

Normal_dropped_calls=Normal_dropped_calls+1

Landmark_colour = green

Else

landmark_colour = red

hand-over_dropped_calls= hand-over_dropped_calls+1

Open(nokia_map)

Plot(gps_coords, landmark)

21. if (total_calls = 10)

Score_hand-over_dropped=

sum(hand-over_dropped_quality)/hand-over_dropped_calls

Score_normal_dropped=

sum(normal_dropped_quality)/normal_dropped_calls

score_successful_attempts=

(sum(hand-over_dropped_quality+sum(normal_dropped_quality))
/total_successful_attempts

22. If (call_attempts_successful< =2) Score_successful_call_rate = 1

Elseif (call_attempts_successful< =3 && average_signal_strength >=4)
Score_successful_call_rate = 2

Elseif (call_attempts_successful< =5 && average_signal_strength >=6)
Score_successful_call_rate =3

Elseif (call_attempts_successful< =7 && average_signal_strength >=8)
Score_successful_call_rate =4

Elseif (_attempts_successful< =9 && average_signal_strength >=10)
Score_successful_call_rate =5

23. If (hand-over_success_calls< 40 per cent) Score_hand-over_success_calls_rate = 1

Elseif (hand-over_success_calls <40 per cent && hand-over_success_calls >60) Score_hand-over_success_calls_rate = 2

Elseif (hand-over_success_calls <60 per cent && hand-over_success_calls >70) Score_hand-over_success_calls_rate =3

Elseif (hand-over_success_calls <70 per cent && hand-over_success_calls >80) Score_hand-over_success_calls_rate =4

Elseif (hand-over_success_calls >80) Score_hand-over_success_calls_rate =5

24. Calculate average_call_quality=(score_successful_attempts+score_successful_call_rate+score_hand-over_success_calls_rate)/3

25. Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful, score_successful_attempts, normal_dropped_calls,score_normal_dropped, hand-over_dropped_calls,score_hand-over_dropped,score_successful_call_rate,score_hand-over_success_calls_rate, average_call_quality)

26. if (sendSMS = auto && whenSMSstat_send = always || sendSMS =auto && whenSMS_call_failed < 5) || whenSMS_hand-over_dropped < 2)

sendSMS(num_calls_unsuccessful,

num_calls_successful,

num_of_calls_dropped_hand-over,

num_normal_dropped)

27. If (log location = internal memory)

save signalmeter.log to c:/data

save calls_stats.log to c:/data

else save signalmeter.log to e:/data

save calls_stats.log to e:/data

28. if(sendSMS = Manual && want_to_send_SMS= yes)

set(mobile_number)

sendSMS(signal_stength, SignalQuality,call_drop_information)

29. End of Program

Table 1 shows the final call quality classification based on the score for a bundle of 10 calls.

Score	Classification
<1	Extremely bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

Table 1: Call quality scores

3. **Bandwidth quality**

The internet bandwidth quality provided by mobile operators fluctuates significantly and frustrates the user sometimes. As the number of users increases in a particular cell of the mobile network, the bandwidth decreases and hence there would be a loss of revenue, if the situation continues for the operator. An attempt to measure the average bandwidth quality per individual and for bundles of 10 downloads is calculated as portrayed in Figure 2.

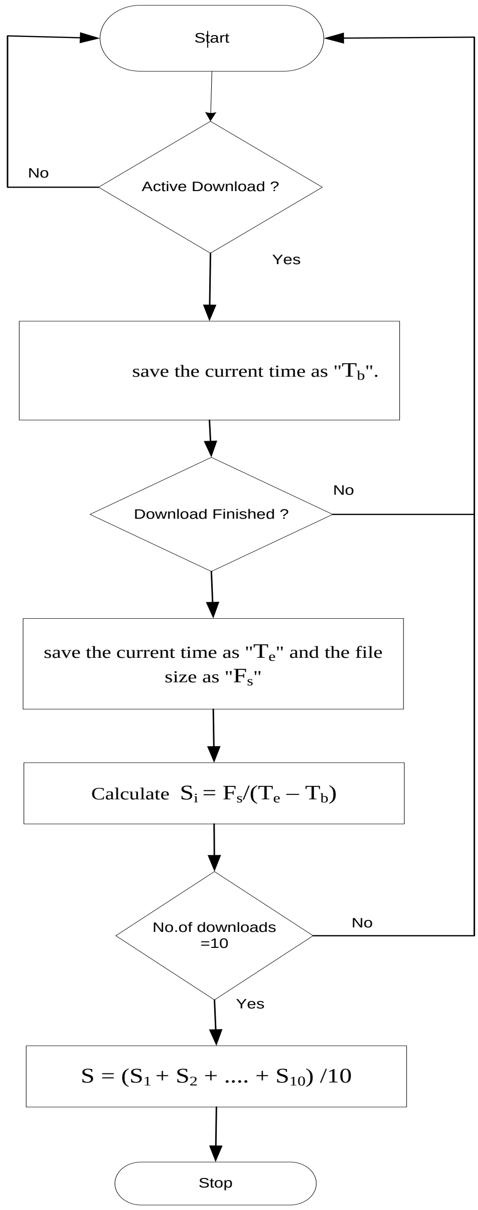

Figure 2: QMeter® bandwidth calculation flow-chart

The scores computed can be used by the user as well as the operator to evaluate bandwidth quality.

The average bandwidth of data download is computed using the following procedure:

- When a new download is initiated save the current time as "T_b".
- When the download is finished save the current time as "T_e" and the file size as "Fs".
- Calculate the average speed for this download as $S_i = F_s/(T_e - T_b)$.

- The average bandwidth for 10 downloads is calculated as $S = (S_1 + S_2 + ... + S_{10}) /10$.

The score for the average bandwidth is computed on the scale of 1-5.

Bandwidth	**Score**
< 32 kbps	1 (Extremely bad)
<32 & <64 kbps	2 (Bad)
<64 & < 128	3 (Average)
<128 & <256	4 (Good)
> 256	5 (Excellent)

Table 2: Bandwidth score

The bandwidth quality scores will give the user and the operator better insight into the usage of bandwidth. The approach can further be enhanced by capturing the cell-id and sending the critical scores for analysis to provide a better service.

4. Conclusion

The research measures call quality and bandwidth quality in mobile telecommunications networks. It can be used to benchmark a mobile network by the user and hence serve as a base for charging the customer by the operators. It can also be used by the telecom authorities to regulate and evaluate mobile operators by regularly checking the network to find out if operators are meeting the required licence criteria of quality of network from the end-user's perspective. Further, it can be used as a consumer protection tool to ensure that tariffs correlate with call quality. The proposed bandwidth quality measurement approach can be used by network operators to enhance the network and provide better service.

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Call Quality Measurement for Telecommunication Network and Proposition of Tariff Rates

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1. Introduction

Traditionally, speech quality was measured offline using subjective listening tests. Subjective testing is called mean opinion score (MOS). It provides a numerical indication of the perceived quality of received human speech. The MOS would be expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality and 5 is the highest. But subjective estimation at various sites within the coverage area is laborious, expensive and time-consuming.

Most systems nowadays use an automatic objective evaluation system to measure speech quality. Objective evaluation is typically achieved by measuring the distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signal power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

Objective testing algorithms are also called automated quality measurement techniques. Three famous objective tests are the perceptual speech quality measure (PSQM), the perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ). Currently, the most popular techniques are those based on psychoacoustics models, referred to as perceptual domain measures [1]. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach [2]. In input-output objective assessment methods, speech quality is estimated by measuring the distortion between an “input” or a reference signal, and an “output” or received signal.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), the modified BSD (MBSD), measuring normalizing blocks

(MNB), the PSQM+, the telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS), and most recently the perceptual evaluation of speech quality (PESQ), which is specified by ITU-T recommendation P. 862 [3], as the international standard for testing networks and codecs. In the case of input-output based speech quality assessment, good correlations were observed, reaching up to 99 per cent in some cases [4]. Correlation between the objective speech quality measure and the subjective quality measure is mostly used as the system (or method) performance measure.

The field of estimating speech quality using only received speech without access to the input record is a relatively new area. In 1994, Jin Liang and R. Kubichek [5] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results, in special cases achieving 90 per cent correlation. R. Kubichek and Chiyi Jin [6] used the vector quantization method, which yields up to 83 per cent correlation.

An output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal is reported in [7]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al.* [8] reported a novel use of vocal-tract modelling techniques, which enable prediction of the quality of a network degraded speech stream to be made in a non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [9]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

Khalid Al-Mashouq and Mohammed Al-Shaye [10] proposed a time-delay multilayer neural network model that can rate speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPCs) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached to 0.87. However, the published result is not generalized due to the limited number of speech files used.

In this paper we propose a new scheme for measuring speech quality over the network from the subscriber set and analyzing the quality in detail. We obtain the signal strength information of an active call every five milliseconds, process signal information and plot landmarks based on an aggregation of successful and unsuccessful calls and their related scores on the map. The critical scores were sent to the relevant authorities and finally new rating schemes were proposed for different quality measures.

2. The experiment

The whole system is divided into four individual processes, namely signal strength measurement, signal strength statistics, plotting landmarks and SMS signal information. The following is a basic outline of our approach in carrying out the experiment in collecting and analyzing the signal quality of an active call.

1. Upon a call being set up the signal level values are taken every five milliseconds.
2. The criteria in Table 1 are used to decide on the quality of the signal.
3. A score is then given to the sample collected every five milliseconds, with five being the best on a scale of 1 to 5 and 1 indicating very bad signal strength.
4. At the end of the call session a cumulative score is computed and based on the score (ranging from 1 to 5) the speech quality is approximately computed.

Signal level range (dBm)	**Classification**
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Signal quality

The signal strength parameters – such as number of normal dropped calls, dropped due to hand-over (cell hand-over), number of failed call attempts (due to network failure), successful call attempts – were also analyzed for a call bundle of 10. The call quality information was plotted as landmarks to visualize the information.

The system works during the active call, which also records the GPS co-ordinates during the call. The system has provision of auto-start which helps the user to log and locate the information of all the active calls on the map, once the auto-start is enabled. This reduces the problem of starting the application every time during an active call. This also helps us to analyze the signal quality at any location for a given operator. It supports both internal and external GPS connected through Bluetooth. The captured GPS co-ordinates were used for plotting the average signal strength on the map. Figure 1 depicts the process flow of plotting landmarks with the measured signal strength information.

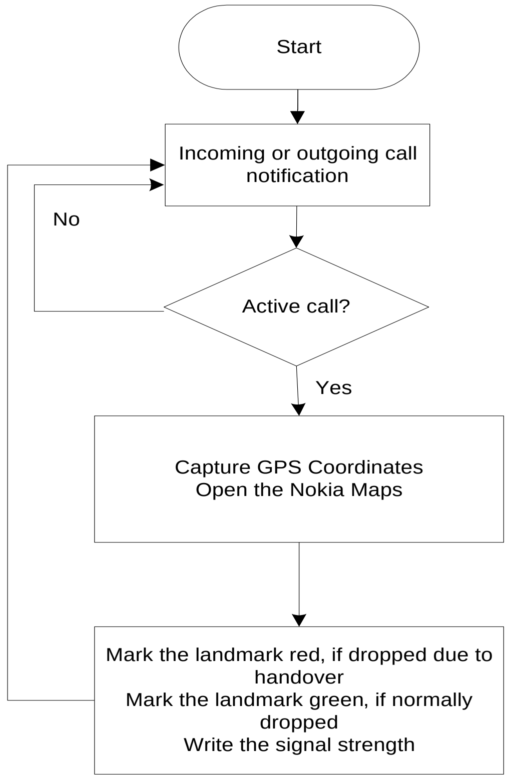

Figure 1: Plotting landmarks

The landmarks marked in red are the calls dropped due to hand-over and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls. The system records the number of successful and unsuccessful call attempts made for every 10 call attempts.

The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The call drop information such as normally dropped from either party or dropped due to hand-over during the cell change is also recorded. The number of normal dropped and hand-over dropped, with their average scores, is also recorded.

The system has the ability to send the signal strength information to the particular number. It can set the mobile number to which the SMS will be sent automatically at the end of the call. The system has the option of setting to send the SMS always, after a less than bad call or based on other criteria. At the end of 10 calls, the call statistics are also sent as an SMS.

3. Charging rate versus quality

New charging rates are proposed based on the four call quality parameters derived in [11]. The variable X is the normal charging rate per minute, n is the number of called minutes in a bundle of 10 call attempts.

Table 2 is the new charging rate proposed, based on the average signal strength of successful calls in a bundle of 10.

Average signal strength of successful calls score	**Charge**
5	X*n
4	X*n
3	X*n*0.75
2	X*n*0.5
1	No charge

Table 2: Proposed charging rate versus average signal strength of successful calls

Table 3 is the new charging rate proposed, based on successful call attempts in a bundle of 10.

Successful call attempts score	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 3: Proposed charging rate versus successful call attempts

Table 4 is the new charging rate proposed, based on average signal strength of normal dropped calls in a bundle of 10.

Normal dropped rate score	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 4: Proposed charging rate versus normal dropped rate

Table 5 is the new charging rate proposed, based on total call quality of calls in a bundle of 10.

Call quality	**Charge**
5(Very good)	X*n
4 (Good)	X*n
3 (Average)	X*n*0.75
2 (Bad)	X*n*0.5
1 (Very bad)	No charge

Table 5: Proposed charging rate versus call quality

4. Conclusion

Analysis of speech quality is major and continuing research that involves investigation of the metrics used to detect the call, obtaining the signal strength information and analyzing the signal information from various perspectives. The proposed method can be used to benchmark the mobile network by the user and hence it can be used as a base for charging the customer by the operators. The charging rates proposed in this work are based on signal quality and the call statistics recorded.

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Emerging Results on Symbian Based Perceptual Evaluation of Speech Quality for Telecommunication Networks

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1. Introduction

The quality of telecommunication voice services has become an important issue due to the evolving and liberalized market. With the advent of new technologies, however, diversification has taken place which makes it necessary to carefully plan and observe network quality. Quality of service is an important factor in gaining customer loyalty, which in turn increases the subscriber base and reduces churn.

Traditionally, a user's perception of speech quality has been measured offline using subjective listening tests. The method of subjective testing called mean opinion score (MOS) provides a numerical indication of the perceived quality of received human speech over the connection. The MOS is expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality and 5 is the highest. But subjective estimation by repeated listening tests at various sites within the coverage area is laborious, expensive and time consuming. It would be much more desirable to use an automatic objective evaluation system that applies a good objective speech quality measure to estimate the statistical average of subjective opinions of the typical conversational speech sentences sent through the mobile network.

Objective testing algorithms are also called automated quality measurement techniques. The three most famous objective methods are:

1. Perceptual speech quality measure (PSQM)
2. Perceptual analysis measurement system (PAMS)
3. Perceptual evaluation of speech quality (PESQ).

The above mentioned perception models for evaluating speech quality were developed independently by Mike Hollier, at BT Labs, and John Beerends, at KPN

Research, on the use of perception for voice quality assessment [1, 2]. Hollier took account not only of the amount, but also the distribution, of audible distortion, which could make quality predictions much more accurate. His work was taken up in 1996 by Antony Rix and forms the core of PAMS [3]. It was not until 1996, following a lengthy international study, that the perceptual models for quality assessment were first standardized. The result of this was that Beerends' model, PSQM, became an ITU-T recommendation (P. 861) for assessing speech codecs [4].

Most available objective assessment techniques are based on an input-output approach [5]. In input-output objective assessment methods, speech quality is estimated by measuring the distortion between an “input” or reference signal, and an “output” or received signal. Using a regression technique, the distortion values are then mapped into estimated quality.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), the modified BSD (MBSD), measuring normalizing blocks (MNB), PSQM+, telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS), and most recently the perceptual evaluation of speech quality (PESQ), which is specified by ITU-T recommendation P. 862 [6], as the international standard for testing networks and codecs. Correlation between the objective speech quality measure and the subjective quality measure is mostly used as the system (or method) performance measure. In the case of input-output based speech quality assessment, good correlations were observed, which reaches up to 99 per cent in some cases [7].

The field of estimating speech quality using only received speech without access to the input record is a relatively new area. Most recently, a couple of attempts to develop more credible non-intrusive speech quality measurements based on perceptual analysis have been reported. In 1994, Jin Liang and R. Kubichek [8] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results in special cases, achieving 90 per cent correlation.

Perceptually-based methods imitate the human listening method, where many parameters and environmental effects are considered. R. Kubichek and Chiyi Jin [9, 10] used the vector quantization method which yields up to 83 per cent correlation. Vector quantization has some disadvantages that yield poor results. One of these is an inherent spectral distortion in representing the actual analysis vector. Since there is only a finite number of code book vectors, the process of choosing the "best" representation of a given spectral vector inherently is equivalent to quantizing the vector and leads, by definition, to a certain level of quantization error. As the size of the code book increases, the quantization error decreases. However, with any finite code book there will always be some non-zero level of quantization error. Furthermore, the storage required for code book vectors is often significant. The larger we make the code book, the more storage space is required for the code book.

Another example of this is an output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal, reported in [11]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al.* [12] reported a novel use of vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in a non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [13]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

In [14] a new output-based speech quality measure, which uses bark spectrum analysis and fifth order perceptual linear prediction (PLP), was introduced. The measure is based on comparing the output speech to an artificial reference signal that is appropriately selected from the optimally clustered reference code book, using the self-organizing map approach coupled with an enhanced k-means technique. The average correlation of this technique reached 0.85 for bark spectrum and 0.61 for PLP coefficient, respectively. Both short duration of speech records and limited number of speakers count as disadvantages of this study. Only two male speakers are used; one for training and the other for testing.

In [15], Khalid Al-Mashouq and Mohammed Al-Shaye proposed a time-delay multilayer neural network model that can rate speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPCs) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached 0.87. However, the published result is not generalized due to the limited number of speech files used.

The latest research was introduced in 2004 [16]. This paper describes a new output-based approach that extends the original pseudo-reference framework by replacing the VQ code book with a discrete hidden Markov model. Correlation between objective and subjective scores shows good performance for text dependent and text independent cases of 0.88 and 0.92 respectively, but does not provide speaker independence. However, none of these output-based investigations are in the mobile communication environment, except in the work of Al-Mashouq and Al-Shaye [17].

Linking a GPS based device with a smart-phone allows the creation of location based intelligence software that has the potential to be infinitely customizable by the user. The software implemented will allow the user to save in “preferences” the user input, which in turn affects the way the software works. The program collects the GPS co-ordinates during the active voice call throughout the movement and associates the signal strength with it. The GPS device may also be outside the phone and contact is made using the Bluetooth protocol. By building the GPS libraries separately, it should be possible to separate the core logic from the communications code using the GPS unit and allow newer versions to function using a built-in GPS resource.

The complexity of this software can be increased whereby the user can use a scripting language to instruct the software what actions need to be carried out depending on their location. This signifies that the software will interpret these scripts and carry out the actions listed, possibly repeated times. A rich user interface can also be constructed which will allow maps to be downloaded and the user location and points of interest in relation to programmed actions to be shown. The software will have the capacity to be flexible by allowing advanced scripting support to logically carry out multiple tasks as well as being able to accept simple commands using an easy-to-use user interface. Security is also important and it should be possible to create a script for the software and disable the ability to remove or modify that script by users without proper authorization. This will be especially beneficial in commercial applications such as tracking and monitoring situations.

In this paper, the results of our ongoing work are presented. This method adopts a non-traditional approach and can be used as a relative index for assessing the quality of speech. Further, the method offers a significant advantage over traditional speech quality measurement as the network operator and subscriber can evaluate speech quality for every call made under different conditions.

The proposed method is part of ongoing research that will involve investigation of metrics which are used to detect the call, obtaining the signal strength information, refreshing the value every five milliseconds and finally implementing the application in a Symbian operating system.

2. The experiment

Our ultimate goal is to develop mobile handset software which will perform an automatic speech quality assessment for every telephone call. It should give an objective score for each call along with a periodic speech quality average. This, in turn, will give subscribers a practical way to assess the performance of different operators.

Our proposed method adopts a non-traditional approach and the aim is to use it as a relative index when assessing the quality of speech. This is applied in the following manner:

1. Upon a call being set up, the signal level values are taken every five milliseconds.
2. The criteria in Table 1 (below) are used to decide on the quality of the signal.
3. A score is then given to the sample, collected every five milliseconds, with five being the best on a scale of 5 and 1 indicating very bad signal strength.

Signal level range (dBm)	Classification
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Perceived quality rating of human speech

At the end of the call session a cumulative score is computed and, based on the score (ranging from 1 to 5), the speech quality is approximately computed.

The signal measure program is developed in developer platform s60 third edition of Symbian. It is user friendly, able to accept the user preference of selecting the input. The program accepts two kinds of settings: log change and log location. In log change, the user can select either the log for every five milliseconds option or when signal strength is changed. The idea behind using these two options is to address the problem of the increasing size of the log file when signal strength is being logged every five milliseconds. In log location, the user can select either internal memory or memory card. An auto-run option, which allows the user to enable and disable the program, is also embedded in the system.

The call quality also works as a location based service using the GPS. The system works during the active call, which also records the GPS co-ordinates during the call. The system has provision of auto-start which helps the user to log and locate the information of all the active calls on the map, once the auto-start is enabled. This reduces the problem of starting the application every time during the active call. It also helps us to analyze the signal quality at any location for a given operator. It supports both internal and external GPS connected through Bluetooth. The captured GPS co-ordinates are used for plotting the average signal strength on the map.

The system also records the number of successful and unsuccessful call attempts made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The call drop information – such as normally dropped from either of the party or dropped due to hand-over during the cell change – is also recorded. The number of normal dropped and hand-over dropped calls, with their average scores, are also recorded. The landmarks marked in red are the calls dropped due to hand-over and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls.

A generic algorithm of our signal measure algorithm is presented below:

Signal_Measure( )

1. Get the preferences for log_change, log_location
2. Get total_calls, Call_attempts_failed, call_attempts_successful, normal_dropped_calls, hand-over_dropped calls
3. if (total_calls =10) reset all variables to zeros
4. if (call_attempt = failed)
        total_calls=total_calls+1
    call_attempts_failed=call_attempts_failed+1
5. While (phone_status != idle && call_attempt = successful)
6. total_calls=total_calls+1
7. call_attempts_successful=call_attempts_successful+1
8. if(gps_coords available)
        Get the gps_coords
9. Get the date, time, cell_id.
10. Get the signal_strength
11. if (log_change= 5ms)
    write("signal_measure.log",date, time, gps_coords, cell_id, signal_strength)
12. if (log_change= when changed) and (signalstrengthchange=yes)
    Write("signal_measure.log",date, time, phone status,signal strength)
13. End of While
14. End of call
15. Calculate average_signal_strength
16. If (average_signal_strength <= -95 && average_signal_strength >= -120)
    SignalQuality= Extremely Bad
    Elseif (average_signal_strength <= -85 &&
    average_signal_strength >-95) SignalQuality=Bad
    Elseif (average_signal_strength <= -75 && average_signal_strength > -85)
    SignalQuality=Average
    Elseif (average_signal_strength <=-65 && average_signal_strength > -75)
    SignalQuality=Good
    Elseif (average_signal_strength <= -55 && average_signal_strength > -65)
    SignalQuality=Very Good
17. Write("signal_measure.log",date, time, phone status, average_ signal_strength)
18. Write (SignalQuality)
19. If (GPS_Coords Available)
    If(call_dropped = Normal)
        Normal_dropped_calls=Normal_dropped_calls+1
        Landmark_colour = green
    Else
        landmark_colour = red
        hand-over_dropped_calls=
        hand-over_dropped_calls+1
    Open(nokia_map)
    Plot(gps_coords, landmark)
20. if (total_calls = 10)
    Score_hand-over_dropped=
    sum(hand-over_dropped_quality)/hand-over_dropped_calls;
    Score_normal_dropped= sum(normal_dropped_quality)/normal_dropped_calls;
    score_successful_attempts=

(sum(hand-over_dropped_quality+sum(normal_dropped_quality)) /total_successful_attempts;

21. Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful, score_successful_attempts, normal_dropped_calls,score_normal_dropped, hand-over_dropped_calls, score_hand-over_dropped)
22. If (log location = internal memory)
    save signalmeter.log to c:/data
    save calls_stats.log to c:/data
    else save signalmeter.log to e:/data
    save calls_stats.log to e:/data
23. End of Program

3. Emerging results

As a step towards realization of our research, the first and second versions of the software have been developed. These have met the following set requirements:

- Measuring signal strength at every five milliseconds during a call and providing average strength of the signal at the end of the call.
- The cumulative scores of normal dropped and dropped due to hand-over recorded every 10 calls.
- The total successful and unsuccessful call attempts recorded for every 10 call attempts.
- The information related to the call is plotted as a landmark on the map, depicting the nature of its drop and the signal strength details.

Figures 1 and 2 depict the landmarks of successful calls, with green and red showing the normally and hand-over dropped calls. The results logged are presented in Appendix 1.

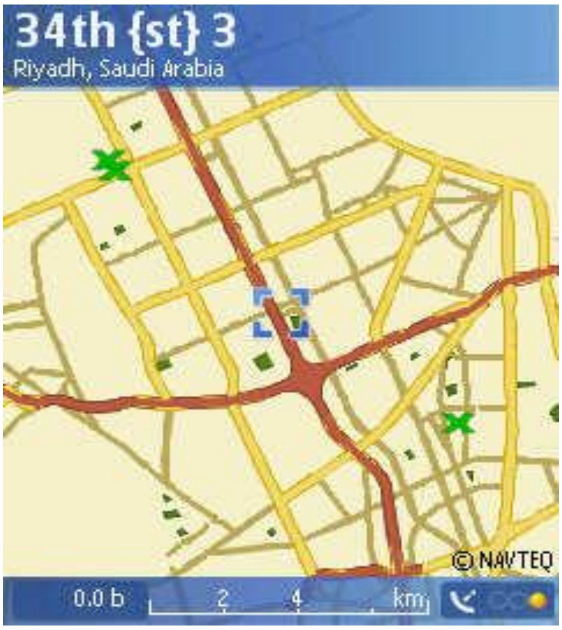

Figure: 1 Landmarks for normally dropped calls

Figure: 2 Landmarks for normally and hand-over dropped calls

Appendix 1: Documentation of operating platform

2009/04/23 - 07:51:21 : : Phone status: Ringing
2009/04/23 - 07:51:35 : : Phone status: Answering
2009/04/23 - 07:51:35 : : Phone status: Connected
2009/04/23 - 07:51:36 : : Current network info LocationAreaCode = 397 CellId = 12843
2009/04/23 - 07:51:36 : : Signal strength is = 91 dBm, 5 bars
2009/04/23 - 07:51:37 : : Signal strength is = 96 dBm, 3 bars
2009/04/23 - 07:51:37 : : Signal strength is = 98 dBm, 2 bars
2009/04/23 - 07:51:38 : : Signal strength is = 97 dBm, 2 bars
2009/04/23 - 07:51:39 : : Signal strength is = 90 dBm, 5 bars
2009/04/23 - 07:51:40 : : Signal strength is = 77 dBm, 7 bars
2009/04/23 - 07:51:40 : : Signal strength is = 81 dBm, 7 bars
2009/04/23 - 07:51:43 : : Signal strength is = 87 dBm, 6 bars
2009/04/23 - 07:51:44 : : Signal strength is = 85 dBm, 6 bars
2009/04/23 - 07:51:44 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2009/04/23 - 07:51:45 : : Average signal strength is 89 dBm (Bad)
2009/04/23 - 07:51:45 : : callInfo etelCode = -65536, netCode = 0
2009/04/23 - 07:51:45 : : ***********************

The calls statistics after 10 calls

2009/04/22 - 15:41:45 : : 0 call attempts failed

2009/04/22 - 15:41:45 : : 10 call attempts successful :: Score: 5 (Very good)
2009/04/22 - 15:41:46 : : 10 calls was normally dropped :: Score: 5 (Very good)
2009/04/22 - 15:41:46 : : ***********************

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Transitional Results on Symbian Based Call Quality Measurement for Telecommunication Network

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1. Introduction

This proposed approach to speech quality measurement in telecommunication networks aims to equip network subscribers with the opportunity to choose their telecom service provider based, among other key indices, upon speech quality. This will also afford the service providers the capability of predicting customers' opinion of quality of service and the need for necessary network optimization for continued customer satisfaction that will ensure loyalty and an increased subscriber base.

Traditionally, speech quality has been measured offline using subjective listening tests. The subjective testing is called mean opinion score (MOS). It provides a numerical indication of the perceived quality of received human speech. The MOS would be expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality, and 5 is the highest. But subjective estimation at various sites within the coverage area is laborious, expensive and time-consuming. Most of the systems nowadays use an automatic objective evaluation system to measure speech quality.

Objective evaluation of speech quality in systems is typically achieved by measuring the distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signal power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

Objective testing algorithms are also called automated quality measurement techniques. Three famous objective tests are the perceptual speech quality measure (PSQM), perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ).

Currently, the most popular techniques are those based on psychoacoustic models, referred to as perceptual domain measures [1]. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach [2]. In input-output objective assessment methods, speech quality is estimated by measuring the distortion between an “input”, or a reference signal, and an “output”, or received signal.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), modified BSD (MBSD), measuring normalizing blocks (MNB), PSQM+, the telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS) and most recently the perceptual evaluation of speech quality (PESQ), which is specified by ITU-T recommendation P. 862 [3] as the international standard for testing networks and codecs. In the case of input-output based speech quality assessment, good correlations were observed, which reaches up to 99 per cent in some cases [4]. Correlation between the objective speech quality measure and the subjective quality measure is mostly used as the system (or method) performance measure.

The field of estimating speech quality using only received speech without access to the input record is a relatively new area. In 1994, Jin Liang and R. Kubichek [5] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results, in special cases achieving 90 per cent correlation. R. Kubichek and Chiyi Jin [6,7] used the vector quantization method, which yields up to 83 per cent correlation.

A hybrid signal-and-link-parametric approach to speech quality measurement for voice-over-internet protocol (VoIP) communications is proposed in [8]. Connection parameters were used to determine a base quality representative of the transmission link. Degradation factors, computed from perceptual features extracted from the decoded speech signal, are used to quantify distortions not captured by the connection parameters. The algorithm is tested on speech degraded by acoustic noise, temporal clippings and noise suppression artifacts, thus simulating degradations present in wireless-VoIP tandem connections.

Speech quality measurement uncertainty over the actual VoIP telephony network is presented in [9]. The uncertainty of the PESQ results under different measurement conditions and real-life VoIP equipment (media gateway) is also analyzed.

An output-based speech quality measure, which uses only the visual effect of a spectrogram of the received speech signal, is reported in [10]. A spectrogram is a two dimensional representation of time-dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al.* [11] reported a novel use of vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [12]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

In [13] a new output-based speech quality measure, which uses bark spectrum analysis and fifth order perceptual linear prediction (PLP) was introduced. The measure is based on comparing the output speech to an artificial reference signal that is appropriately selected from optimally clustered reference code book, using the self-organizing map approach coupled with an enhanced k-means technique. The average correlation of this technique reached 0.85 for bark spectrum and 0.61 for PLP coefficient, respectively. Both short duration of speech records and limited number of speakers count as disadvantages of this study. Only two male speakers were used; one for training and the other one for testing.

Khalid Al-Mashouq and Mohammed Al-Shaye, in [14], proposed a time-delay multilayer neural network model that can rate speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPCs) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent cases reached 0.87. However, the published result is not generalized due to the limited number of speech files used.

In [15] the latest output-based approach that extends the original pseudo-reference framework by replacing the VQ code book with a discrete hidden Markov model was proposed. Correlation between objective and subjective scores shows good performance for text dependent and text independent cases, of 0.88 and 0.92 respectively, but does not provide speaker independence. None of the output-based investigations belong to the mobile communication environment, except the work of Al-Mashouq and Al-Shaye [16].

In this paper we propose a new scheme which is part of ongoing research for measuring speech quality over the network from the subscriber set. This method adopts a non-traditional approach and can be used as a relative index for assessing the quality of speech. Further, the method offers a significant advantage over traditional speech quality measurement as the network operator and subscriber can evaluate speech quality for every call made under different conditions

The proposed method is part of ongoing research that will involve investigating the metrics that are used to detect the call, obtaining the signal strength information, refreshing the value at every five milliseconds, signal information processing and plotting based on aggregation of successful and unsuccessful calls and their related scores on the map. Finally, the application will be implemented in a Symbian operating system.

2. The experiment

As a starting point, as stated earlier, our proposed method adopts a non-traditional approach and the aim is to use it as a relative index when assessing the quality of speech. This is applied in the following manner:

1. Upon a call being set up, the signal level values are taken every five milliseconds.
2. The criteria in Table 1 (below) are used to decide on the quality of the signal.
3. A score is then given to the sample, collected every five milliseconds, with five being the best on a scale of 5 and 1 indicating very bad signal strength.

At the end of the call session a cumulative score is computed and, based on the score (ranging from 1 to 5), the speech quality is approximately computed. The following are the classification criteria for signal strength:

Signal level range (dBm)	**Classification**
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Perceived quality rating of human speech

The signal measure program is developed in the developer platform s60 third edition of Symbian. It is user friendly, able to accept the user preference of selecting the input. The program accepts two kinds of settings: log change and log location. In log change, the user can select either the log for every five milliseconds option or when signal strength is changed. The idea behind using these two options is to address the problem of the increasing size of the log file when signal strength is being logged every five milliseconds. In log location, the user can select either internal memory or memory card. An auto-run option, which allows the user to enable and disable the program, is also embedded in the system.

Call quality also works as a location based service using GPS. The system works during the active call, which also records the GPS co-ordinates during the call. The system has provision of auto-start which helps the user to log and locate the information of all the active calls on the map, once the auto-start is enabled. This reduces the problem of starting the application every time during the active call. It also helps us to analyze the signal quality at any location for a given operator. It supports both internal and external GPS connected through Bluetooth. The captured GPS co-ordinates are used for plotting the average signal strength on the map.

The system also records the number of successful and unsuccessful call attempts made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The call drop information – such as normally dropped from either of the party or dropped due to hand-over during the cell change – is also recorded. The number of normal dropped and hand-over dropped calls, with their average scores, also recorded. The landmarks marked in red are the calls dropped due to hand-over and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls.

A generic algorithm of our signal measure algorithm is presented below:

Signal_Measure( )

1. Get the parameters of log_change, log_location
2. Get total_calls, Call_attempts_failed, call_attempts_successfulA call quality meter has a built-in module, where the call quality escalation has been addressed from the end-user perspective as an SMS sending to the msisdn which can be saved by the user, normal_dropped_calls, hand-over_dropped calls
3. if (total_calls =10) reset all variables to zeros
4. if (call_attempt = failed)
   total_calls=total_calls+1
   call_attempts_failed=call_attempts_failed+1
5. While (phone_status != idle && call_attempt = successful)
6. total_calls=total_calls+1
7. call_attempts_successful=call_attempts_successful+1
8. if(gps_coords available)
   Get the gps_coords
9. Get the date, time, cell_id.
10. Get the signal_strength
11. if (log_change= 5ms)
    write(“signal_measure.log”,date, time, gps_coords, cell_id, signal_strength)
12. if (log_change= when changed) and (signalstrengthchange=yes)
    Write(“signal_measure.log”,date, time, phone status,signal strength)
13. End of While
14. End of call
15. Calculate average_signal_strength
16. If (average_signal_strength <= -95 && average_signal_strength >= -120)
    SignalQuality= Extremely Bad
    Elseif (average_signal_strength <= -85 &&
    average_signal_strength >-95) SignalQuality=Bad
    Elseif (average_signal_strength <= -75 && average_signal_strength > -85)
    SignalQuality=Average
    Elseif (average_signal_strength <=-65 && average_signal_strength > -75)
    SignalQuality=Good
    Elseif (average_signal_strength <= -55 && average_signal_strength > -65)
    SignalQuality=Very Good
17. Write(“signal_measure.log”,date, time, phone status, average_ signal_strength)

18. Write (SignalQuality)
19. If (GPS_Coords Available)
    If(call_dropped = Normal)
      Normal_dropped_calls=Normal_dropped_calls+1
        Landmark_colour = green
    Else
        landmark_colour = red
        hand-over_dropped_calls=
        hand-over_dropped_calls+1
    Open(nokia_map)
    Plot(gps_coords, landmark)
20. if (total_calls = 10)
    Score_hand-over_dropped=
    sum(hand-over_dropped_quality)/hand-over_dropped_calls;
    Score_normal_dropped= sum(normal_dropped_quality)/normal_dropped_calls;
    score_successful_attempts=
    (sum(hand-over_dropped_quality+sum(normal_dropped_quality))
    /total_successful_attempts;
21. Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful, score_successful_attempts, normal_dropped_calls,score_normal_dropped, hand-over_dropped_calls, score_hand-over_dropped)
22. If (log location = internal memory)
    save signalmeter.log to c:/data
    save calls_stats.log to c:/data
    else save signalmeter.log to e:/data
    save calls_stats.log to e:/data
23. End of Program

3. Results and analysis

As a step towards realization of our research, the first and second version of the software has been developed. This has met the following set requirements:

- Measuring signal strength at every five milliseconds during a call and providing average strength of the signal at the end of the call.
- The cumulative scores of normal dropped and dropped due to hand-over recorded every 10 calls.
- The total successful and unsuccessful call attempts recorded for every 10 call attempts.
- The information related to the call is plotted as a landmark on the map, depicting the nature of its drop and the signal strength details.

Figures 1 and 2 depict the landmarks of successful calls, with green and red showing the normally and hand-over dropped calls. The results logged are presented in Appendix 1.

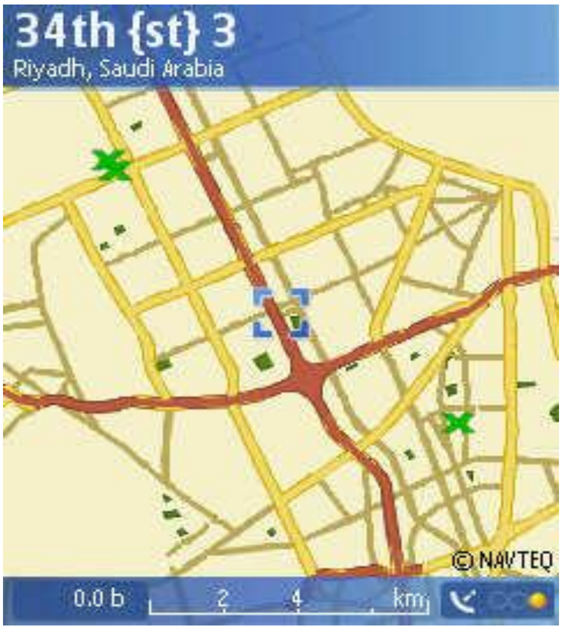

Figure: 1 Landmarks for normally dropped calls

Figure: 2 Landmarks for normally and hand-over dropped calls

Appendix 1: Documentation of operating platform

2009/05/21 - 13:20:35 : : Current network info LocationAreaCode = 352 CellId = 12211
2009/05/21 - 13:20:36 : : Signal strength is = 80 dBm, 7 bars
2009/05/21 - 13:20:58 : : Signal strength is = 83 dBm, 7 bars
2009/05/21 - 13:20:59 : : Signal strength is = 82 dBm, 7 bars
2009/05/21 - 13:21:07 : : Signal strength is = 77 dBm, 7 bars

2009/05/21 - 13:21:12 : : Signal strength is = 81 dBm, 7 bars
2009/05/21 - 13:21:44 : : Signal strength is = 79 dBm, 7 bars
2009/05/21 - 13:21:46 : : Signal strength is = 82 dBm, 7 bars
2009/05/21 - 13:21:47 : : Signal strength is = 78 dBm, 7 bars
2009/05/21 - 13:21:49 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2009/05/21 - 13:21:49 : : Average signal strength is 80 dBm (Average)

Sample call statistics

2009/05/24 - 07:45:33 : : 0 call attempts failed
2009/05/24 - 07:45:33 : : 10 call attempts successful :: Score: 3 (Average)
2009/05/24 - 07:45:33 : : 10 calls was normally dropped :: Score: 3 (Average)
2009/05/24 - 07:45:33 : : **********************

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Perceptual Evaluation of Call Quality Measurement Using a Non-Traditional Symbian Operating System

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1. Introduction

Traditionally, a user's perception of speech quality is measured offline using subjective listening tests. The method of subjective testing, called mean opinion score (MOS), provides a numerical indication of the perceived quality of received human speech over the connection. The MOS is expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality and 5 is the highest. But subjective estimation by repeated listening tests at various sites within the coverage area is laborious, expensive and time-consuming. It would therefore be much more desirable to use an automatic objective evaluation system that applies a good objective speech quality measure to estimate the statistical average of subjective opinions of the typical conversational speech sentences sent through the mobile network.

Objective assessment of speech quality in modern communication systems is typically achieved by measuring some form of distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signals power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

The results of MOS testing are expensive and impractical for testing in the field. Automatic testing algorithms were created in an attempt to formulate objective network testing similar to signal to noise ratio (SNR), bit error rate (BER) and received signal strength (RSSI), which are used to measure the signal quality. Objective testing algorithms are also called automated quality measurement techniques. Three objective tests were developed:

1. Perceptual speech quality measure (PSQM)
2. Perceptual analysis measurement system (PAMS)
3. Perceptual evaluation of speech quality (PESQ).

Perception models for evaluating speech quality were developed independently by Mike Hollier, at BT Labs, and John Beerends, of KPN Research. The latter led subsequent innovations in the 1990s on the use of perception for voice quality assessment [1, 2]. Hollier observed that taking account not just of the amount, but also the distribution, of audible distortion could make quality predictions much more accurate. His work was taken up in 1996 by Antony Rix and forms the core of PAMS [3]. It was not until 1996, following a lengthy international study, that the perceptual model for quality assessment was first standardized. The result of this was that Beerends' model, PSQM, became an ITU-T recommendation (P. 861) for assessing speech codecs [4].

Over the last decade, researchers and engineers in the field of objective measures of speech quality have developed different techniques based on various speech analysis models. Currently, the most popular techniques are those based on psychoacoustics models, referred to as perceptual domain measures [5]. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach [6]. In input-output objective assessment methods, speech quality is estimated by measuring the distortion between an “input”, or a reference signal, and an “output”, or received signal. Using a regression technique, the distortion values are then mapped into estimated quality.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), modified BSD (MBSD), measuring normalizing blocks (MNB), PSQM+, the telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS), and most recently the perceptual evaluation of speech quality (PESQ), which is specified by ITU-T recommendation P. 862 [7], as the international standard for testing networks and codecs. In the case of input-output based speech quality assessment, good correlations were observed, reaching up to 99 per cent in some cases [8].

The field of estimating the speech quality using only received speech without access to the input record is relatively new. Most recently, a couple of attempts to develop more credible non-intrusive speech quality measurements based on perceptual analysis have been reported. In 1994, Jin Liang and R. Kubichek [9] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results, in special cases achieving 90 per cent correlation.

Perceptually-based methods imitate the human listening method where many parameters and environmental effects are considered. R. Kubichek and Chiyi Jin [10, 11] used the vector quantization method, which yields up to 83 per cent correlation.

Vector quantization has some disadvantages that yield poor results. One of these disadvantages is an inherent spectral distortion in representing the actual analysis vector. Since there is only a finite number of code book vectors, the process of choosing the "best" representation of a given spectral vector inherently is equivalent to quantizing the vector. It leads, by definition, to a certain level of quantization error. As the size of the code book increases, the quantization error decreases. However, with any finite code book there will always be some non-zero level of quantization error. Furthermore, the storage required for code book vectors is often significant. The larger we make the code book, the more storage space is required for the code book entries.

Another example of this is an output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal, reported in [12]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al.* [13] reported a novel use of vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in a non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [14]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

In [15] a new output-based speech quality measure, which uses bark spectrum analysis and fifth order perceptual linear prediction PLP, was introduced. The measure is based on comparing the output speech to an artificial reference signal that is appropriately selected from optimally clustered reference code book, using the self-organizing map approach coupled with an enhanced k-means technique. The average correlation of this technique reached 0.85 for bark spectrum and 0.61 for PLP coefficient, respectively. Both short duration of speech records and limited number of speakers count as disadvantages in this study. Only two male speakers were used; one for training and the other one for testing.

In [16], Khalid Al-Mashouq and Mohammed Al-Shaye proposed a time-delay multilayer neural network model that can rate the speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPCs) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached 0.87. However, the published result is not generalized due to the limited number of speech files used.

The latest research was introduced in 2004 [17]. This paper describes a new output-based approach that extends the original pseudo-reference framework by replacing the VQ code book with a discrete hidden Markov model. Correlation between objective and subjective scores shows good performance for text dependent and text independent cases of 0.88 and 0.92 respectively, but does not provide speaker independence. However, the only research in the mobile communication environment is by Al-Mashouq and Al-Shaye [18].

In this work we propose a new scheme for measuring speech quality over the network from the subscriber set. This method adopts a non-traditional approach and can be used as a relative index for assessing the quality of speech. Further, the method offers a significant advantage over traditional speech quality measurement as the network operator and subscriber can evaluate the speech quality for every call made under different conditions. This can be applied in a number of ways:

1. In present day multi-operator environments, service subscribers can easily use this method to choose between different networks.
2. Telecom regulatory bodies can apply this method to enact laws that will make operators charge less for less voice quality measured from their networks.
3. Network operators can use this method as a marketing tool and offer subscribers reduced tariffs whenever the speech quality of calls they make is lower than a particular threshold.

The proposed method is part of ongoing research that will involve investigation of metrics which are used to detect the call, obtaining the signal strength information, refreshing the value at every five milliseconds and finally implementing the application in a Symbian operating system.

2. The proposed method

Our ultimate goal in this work is to develop mobile handset software which will perform an automatic speech quality assessment for every telephone call. It should give an objective score for each call along with a periodic speech quality average. This, in turn, will give subscribers a practical way to assess the performance of different operators. Moreover, operators may want to give their employees a version of this software to do internal network auditing.

As a starting point, we collected a cellular speech database and its original version. Then, we built up a robust continuous speech recognizer. Finally, we derived an automatic machine score wherein correlation with its input-output counterpart is high. The speech databases used in most of the simulation stage are the well-known TIMIT and CTIMIT database set. CTIMIT is the cellular version of the TIMIT phonetic database. We chose these because of their large vocabulary and mixed speakers' accents. The hidden Markov models have to be trained using a clean speech database. The noisy speech database is used for testing. Each word of the database is segmented and phonemes assigned. Then that word is trained for the hidden Markov model parameters using the Baum-Welch algorithm.

For test word recognition, we used the forward or backward algorithm of the hidden Markov model to find the most likely word in the database. The recognized word's probability is categorized on 1-5 scale of quality. Then the test scores are correlated with the PESQ algorithm.

As stated earlier, our proposed method adapts a non-traditional approach and the aim is to use it as a relative index when assessing the quality of speech. This is applied in the following manner:

1. Upon a call being set up, the signal level values are taken every five milliseconds.
2. The criteria in Table 1 (below) are used to decide on the quality of the signal.
3. A score is then given to the sample collected every five milliseconds; with five being the best and 1 indicating very bad signal strength.
4. At the end of the call session a cumulative score is computed. Based on the score (ranging from 1 to 5) the speech quality is approximately computed.

Signal level range (dBm)	**Classification**
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Perceived quality rating of human speech

3. Emerging results

As stated earlier, our proposed method adopts a non-traditional approach and the aim is to use it as a relative index when assessing quality of speech. As a step towards the realization of our research object, the first and second versions of the software have been developed. These have met our set requirement of measuring signal strength every five milliseconds during a call and provide average strength of the signal at the end of the call. An excerpt of documentation of the first and second phase of the operating platform is shown in Appendix 1.

4. Summary

This report has examined a more liberal approach to speech quality measurement in telecommunication networks. It aims to equip the network subscriber with the opportunity to choose his or her telecom service provider based, among other key indices, upon speech quality. This will also afford service providers the capability of predicting customers' opinion of quality of service and the need for network optimization for continued customer satisfaction that will ensure loyalty and an increased subscriber base.

Appendix 1: Documentation of operating platform

2008/05/01 - 15:08:41 : : Current network info LocationAreaCode = 3341 CellId = 6826328
2008/05/01 - 15:08:41 : : Signal strength is = - 95 Db
2008/05/01 - 15:09:09 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/01 - 15:09:11 : : Current network info LocationAreaCode = 3341 CellId = 6827960
2008/05/01 - 15:09:11 : : Call drop observer -> Event : The current cell is changed
2008/05/01 - 15:09:11 : : Call drop observer -> Your call was dropped due to Hand-over
2008/05/01 - 15:09:15 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/01 - 15:09:15 : : Average signal strength is
2008/05/01 - 15:09:15 : : **********************
2008/05/01 - 15:22:37 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/01 - 15:22:41 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/01 - 15:22:41 : : **********************
2008/05/26 - 05:30:58 : : Current network info LocationAreaCode = 3341 CellId = 6827960
2008/05/26 - 05:30:58 : : Signal strength is = - 52 Db
2008/05/26 - 05:31:22 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/26 - 05:31:22 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/26 - 05:31:22 : : Average signal strength is - 52 Db (Very good)
2008/05/26 - 05:31:22 : : **********************
2008/05/26 - 05:34:03 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/26 - 05:34:03 : : **********************
2008/05/26 - 05:34:16 : : Current network info LocationAreaCode = 3341 CellId = 6827960
2008/05/26 - 05:34:16 : : Signal strength is = - 52 Db
2008/05/26 - 05:34:45 : : GPS coordinates lat = 24.7078 , long = 46.6619 , accuracy = 335.95 m, altitude = 612.0000 m , altitude accuracy = 109.50 m
2008/05/26 - 05:34:47 : : GPS coordinates lat = 24.7078 , long = 46.6614 , accuracy = 223.00 m, altitude = 612.0000 m , altitude accuracy = 100.00 m
2008/05/26 - 05:34:48 : : GPS coordinates lat = 24.7078 , long = 46.6616 , accuracy = 281.89 m, altitude = 612.0000 m , altitude accuracy = 89.50 m
2008/05/26 - 05:34:49 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/26 - 05:34:50 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/26 - 05:34:50 : : Average signal strength is - 52 Db (Very good)
2008/05/26 - 05:34:50 : : **********************
2008/05/26 - 05:34:50 : : GPS coordinates lat = 24.7077 , long = 46.6616 , accuracy = 315.13 m, altitude = 612.0000 m , altitude accuracy = 80.00 m
2008/05/26 - 05:34:51 : : GPS coordinates lat = 24.7078 , long = 46.6615 , accuracy = 290.06 m, altitude = 612.0000 m , altitude accuracy = 71.50

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Results of Ongoing Symbian Based Call Quality Measurement for Telecommunication Network

Akram Aburas, Professor J. G. Gardiner, Professor Khalid Al-Mashouq and

Dr Zeyad Al Hokail

1. Introduction

The proposed approach to speech quality measurement in telecommunication networks aims to equip network subscribers with the opportunity to choose their telecom service provider based, among other key indices, upon speech quality. This will also afford service providers the capability of predicting customers' opinion of quality of service and the need for network optimization for continued customer satisfaction that will ensure loyalty and an increased subscriber base.

Traditionally, speech quality has been measured offline using subjective listening tests. The subjective testing is called mean opinion score (MOS). It provides a numerical indication of the perceived quality of received human speech. The MOS would be expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality, and 5 is the highest. But subjective estimation at various sites within the coverage area is laborious, expensive and time-consuming. Most of the systems nowadays use an automatic objective evaluation system to measure the speech quality.

Objective evaluation of speech quality in systems is typically achieved by measuring the distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signals power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

Objective testing algorithms are also called automated quality measurement techniques. Three objective tests are the perceptual speech quality measure (PSQM), the perceptual analysis measurement system (PAMS) and perceptual evaluation of speech quality (PESQ).

Currently, the most popular techniques are those based on psychoacoustics models, referred to as perceptual domain measures. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach. In input-output objective assessment methods, the speech quality is estimated by measuring the distortion between an "input" or a reference signal and an "output" or received signal.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), modified BSD (MBSD), measuring normalizing blocks (MNB), PSQM+, the telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS), and most recently the perceptual evaluation of speech quality (PESQ), which is specified by ITU-T recommendation P. 862 [1], as the international standard for testing networks and codecs. In the case of input-output based speech quality assessment, good correlations were observed, which reaches up to 99 per cent in some cases.

The field of estimating the speech quality using only received speech without access to the input record is a relatively new area. In 1994, Jin Liang and R. Kubichek [2] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results in special cases, achieving 90 per cent correlation. R. Kubichek and Chiyi Jin [3,4] used the vector quantization method, which yields up to 83 per cent correlation.

A hybrid signal-and-link-parametric approach to speech quality measurement for voice-over-internet protocol (VoIP) communications is proposed in [5]. Connection parameters are used to determine a base quality representative of the transmission link. Degradation factors, computed from perceptual features extracted from the decoded speech signal, are used to quantify distortions not captured by the connection parameters. The algorithm is tested on speech degraded by acoustic noise, temporal clippings and noise suppression artefacts, thus simulating degradations present in wireless-VoIP tandem connections.

Another example of this is an output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal, reported in [6]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al*. [7] reported a novel use of vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in a non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [8]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

In [9] a new output-based speech quality measure, which uses bark spectrum analysis and fifth order perceptual linear prediction PLP, was introduced. The measure is based on comparing the output speech to an artificial reference signal that is appropriately selected from optimally clustered reference code book, using the self-organizing map approach coupled with an enhanced k-means technique. The average correlation of this technique reached 0.85 for bark spectrum and 0.61 for PLP coefficient, respectively. Both short duration of speech records and limited number of speakers count as disadvantages in this study. Only two male speakers were used; one for training and the other one for testing.

In [10], Khalid Al-Mashouq and Mohammed Al-Shaye proposed a time-delay multilayer neural network model that can rate the speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPCs) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached 0.87. However, the published result is not generalized due to the limited number of speech files used.

The latest research was introduced in 2004 [11]. This paper describes a new output-based approach that extends the original pseudo-reference framework by replacing the VQ code book with a discrete hidden Markov model. Correlation between objective and subjective scores shows good performance for text dependent and text independent cases of 0.88 and 0.92 respectively, but does not provide speaker independence. However, the only research in the mobile communication environment is by Al-Mashouq and Al-Shaye [12].

In this paper we propose a new scheme which is part of the ongoing research for measuring speech quality over the network from the subscriber set. This method adopts a non-traditional approach and can be used as a relative index for assessing the quality of speech. Further, the method offers a significant advantage over traditional speech quality measurement as the network operator and subscriber can evaluate the speech quality for every call made under different conditions

The proposed method is part of ongoing research that will involve investigation of metrics used to detect the call, obtain the signal strength information, refresh the value every five milliseconds, process signal information and plot based on an aggregation of successful and unsuccessful calls and their related scores on the map. Finally, the application will be implemented in the Symbian operating system.

2. The experiment

As a starting point, as stated earlier, our proposed method adopts a non-traditional approach and the aim is to use it as a relative index when assessing the quality of speech. This is applied in the following manner:

1. Upon a call being set up, the signal level values are taken every five milliseconds.
2. The criteria in Table 1 (below) are used to decide on the quality of the signal.
3. A score is then given to the sample, collected every five milliseconds, with five being the best on a scale of 5 and 1 indicating very bad signal strength.

At the end of the call session a cumulative score is computed and, based on the score (ranging from 1 to 5), the speech quality is approximately computed. The following are the classification criteria for signal strength:

Signal level range (dBm)	Classification
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Perceived quality rating of human speech

The signal measure program is developed in developer platform s60 third edition of Symbian. It is user friendly, able to accept the user preference of selecting the input. The program accepts two kinds of settings: log change and log location. In log change, the user can select either the log for every five milliseconds option or when signal strength is changed. The idea behind using these two options is to address the problem of the increasing size of the log file when signal strength is being logged every five milliseconds. In log location, the user can select either internal memory or memory card. An auto-run option, which allows the user to enable and disable the program, is also embedded in the system.

Call quality also works as a location based service using GPS. The system works during the active call, which also records the GPS co-ordinates during the call. The system has provision of auto-start which helps the user to log and locate the information of all the active calls on the map, once the auto-start is enabled. This reduces the problem of starting the application every time during the active call. It also helps us to analyze the signal quality at any location for a given operator. It supports both internal and external GPS connected through Bluetooth. The captured GPS co-ordinates are used for plotting the average signal strength on the map.

The system also records the number of successful and unsuccessful call attempts made for every 10 call attempts. The successful and unsuccessful call attempts are classified based on whether the call is successfully connected by the network. The call drop information – such as normally dropped from either of the party or dropped due to hand-over during the cell change – is also recorded. The number of normal dropped and hand-over dropped with their average scores also recorded. The landmarks marked in red are the calls dropped due to hand-over and the landmarks marked in green are normally dropped calls. The different colours help one to easily visualize and analyze the calls.

A generic algorithm of our signal measure algorithm is presented below:

Signal_Measure( )

1. Get the parameters of log_change, log_location

2. Get total_calls, Call_attempts_failed, call_attempts_successful, normal_dropped_calls, hand-over_dropped calls
3. if (total_calls =10) reset all variables to zeros
4. if (call_attempt = failed)
    total_calls=total_calls+1
    call_attempts_failed=call_attempts_failed+1
5. While (phone_status != idle && call_attempt = successful)
6. total_calls=total_calls+1
7. call_attempts_successful=call_attempts_successful+1
8. if(gps_coords available)
    Get the gps_coords
9. Get the date, time, cell_id.
10. Get the signal_strength
11. if (log_change= 5ms)
    write("signal_measure.log",date, time, gps_coords, cell_id, signal_strength)
12. if (log_change= when changed) and (signalstrengthchange=yes)
    Write("signal_measure.log",date, time, phone status,signal strength)
13. End of While
14. End of call
15. Calculate average_signal_strength
16. If (average_signal_strength <= -95 && average_signal_strength >= -120)
    SignalQuality= Extremely Bad
    Elseif (average_signal_strength <= -85 &&
    average_signal_strength >-95) SignalQuality=Bad
    Elseif (average_signal_strength <= -75 && average_signal_strength > -85)
    SignalQuality=Average
    Elseif (average_signal_strength <=-65 && average_signal_strength > -75)
    SignalQuality=Good
    Elseif (average_signal_strength <= -55 && average_signal_strength > -65)
    SignalQuality=Very Good
17. Write("signal_measure.log",date, time, phone status, average_ signal_strength)
18. Write (SignalQuality)
19. If (GPS_Coords Available)
    If(call_dropped = Normal)
        Normal_dropped_calls=Normal_dropped_calls+1
        Landmark_colour = green
    Else
        landmark_colour = red
        hand-over_dropped_calls=
        hand-over_dropped_calls+1
    Open(nokia_map)
    Plot(gps_coords, landmark)
20. if (total_calls = 10)
    Score_hand-over_dropped=
    sum(hand-over_dropped_quality)/hand-over_dropped_calls;
    Score_normal_dropped= sum(normal_dropped_quality)/normal_dropped_calls;
    score_successful_attempts=
    (sum(hand-over_dropped_quality+sum(normal_dropped_quality))
    /total_successful_attempts;
21. Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful,

score_successful_attempts, normal_dropped_calls,score_normal_dropped, hand-over_dropped_calls, score_hand-over_dropped)

22. If (log location = internal memory)
    save signalmeter.log to c:/data
    save calls_stats.log to c:/data
    else save signalmeter.log to e:/data
    save calls_stats.log to e:/data
23. End of Program

3. Results and analysis

As a step towards realization of our research, the first and second versions of the software have been developed. These have met the following set requirements:

- Measuring signal strength at every five milliseconds during a call and providing average strength of the signal at the end of the call.
- The cumulative scores of normal dropped and dropped due to hand-over recorded every 10 calls.
- The total successful and unsuccessful call attempts recorded for every 10 call attempts.
- The information related to the call is plotted as a landmark on the map, depicting the nature of its drop and the signal strength details.

Figures 1 and 2 depict the landmarks of successful calls, with green and red showing the normally and hand-over dropped calls. The results logged are presented in Appendix 1.

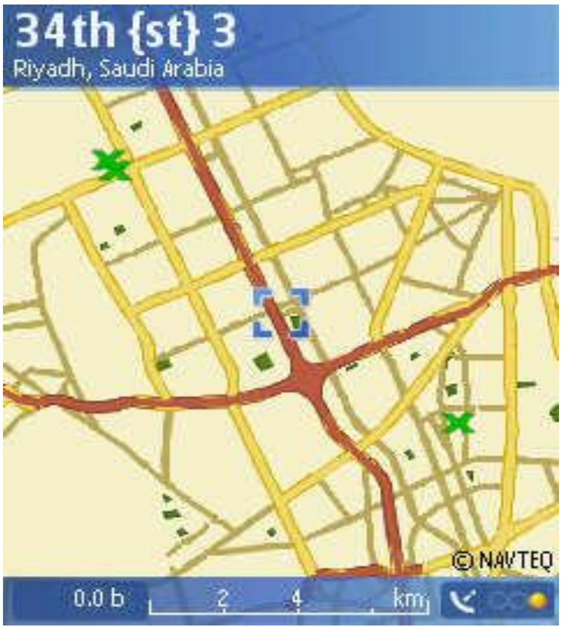

Figure: 1 Landmarks for normally dropped calls

Figure: 2 Landmarks for normally and hand-over dropped calls

Appendix 1: Documentation of operating platform

2009/05/21 - 13:20:35 : : Current network info LocationAreaCode = 352 CellId = 12211
2009/05/21 - 13:20:36 : : Signal strength is = 80 dBm, 7 bars
2009/05/21 - 13:20:58 : : Signal strength is = 83 dBm, 7 bars
2009/05/21 - 13:20:59 : : Signal strength is = 82 dBm, 7 bars
2009/05/21 - 13:21:07 : : Signal strength is = 77 dBm, 7 bars
2009/05/21 - 13:21:12 : : Signal strength is = 81 dBm, 7 bars
2009/05/21 - 13:21:44 : : Signal strength is = 79 dBm, 7 bars
2009/05/21 - 13:21:46 : : Signal strength is = 82 dBm, 7 bars
2009/05/21 - 13:21:47 : : Signal strength is = 78 dBm, 7 bars
2009/05/21 - 13:21:49 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2009/05/21 - 13:21:49 : : Average signal strength is 80 dBm (Average)

Sample call statistics

2009/05/24 - 07:45:33 : : 0 call attempts failed
2009/05/24 - 07:45:33 : : 10 call attempts successful :: Score: 3 (Average)
2009/05/24 - 07:45:33 : : 10 calls was normally dropped :: Score: 3 (Average)
2009/05/24 - 07:45:33 : : ***********************

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Symbian Based Perceptual Evaluation of Speech Quality for Telecommunication Networks

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1. **Introduction**

The quality of telecommunication voice services has become an important issue due to the evolving and liberalized market. A characteristic of the competitive telecommunications market is the convergence in many aspects of service, such as price, between different operators. In this environment, speech quality is becoming a key factor distinguishing one operator from another – a strong indicator of customer satisfaction. Therefore, it is imperative that a service provider is capable of predicting customers' perceptions of quality so that networks can be optimized and maintained.

Traditionally, a user's perception of speech quality is measured offline using subjective listening tests. The method of subjective testing, called mean opinion score (MOS), provides a numerical indication of the perceived quality of received human speech over the connection. The MOS is expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality and 5 is the highest. But subjective estimation by repeated listening tests at various sites within the coverage area is laborious, expensive and time-consuming. It would therefore be much more desirable to use an automatic objective evaluation system that applies a good objective speech quality measure to estimate the statistical average of subjective opinions of the typical conversational speech sentences sent through the mobile network.

Objective assessment of speech quality in modern communication systems is typically achieved by measuring some form of distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signals power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

The results of MOS testing are expensive and impractical for testing in the field.

Automatic testing algorithms were created in an attempt to formulate objective network testing similar to signal to noise ratio (SNR), bit error rate (BER) and received signal strength (RSSI), which are used to measure the signal quality. Objective testing algorithms are also called automated quality measurement techniques. Three objective tests were developed:

1. Perceptual speech quality measure (PSQM)
2. Perceptual analysis measurement system (PAMS)
3. Perceptual evaluation of speech quality (PESQ).

Perception models for evaluating speech quality were developed independently by Mike Hollier, at BT Labs, and John Beerends, of KPN Research. The latter led subsequent innovations in the 1990s on the use of perception for voice quality assessment [1, 2]. Hollier observed that taking account not just of the amount, but also the distribution, of audible distortion could make quality predictions much more accurate. His work was taken up in 1996 by Antony Rix and forms the core of PAMS [3]. It was not until 1996, following a lengthy international study, that the perceptual model for quality assessment was first standardized. The result of this was that Beerends' model, PSQM, became an ITU-T recommendation (P. 861) for assessing speech codecs [4].

Over the last decade, researchers and engineers in the field of objective measures of speech quality have developed different techniques based on various speech analysis models. Currently, the most popular techniques are those based on psychoacoustics models, referred to as perceptual domain measures [5]. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach [6]. In input-output objective assessment methods, speech quality is estimated by measuring the distortion between an “input”, or a reference signal, and an “output”, or received signal. Using a regression technique, the distortion values are then mapped into estimated quality.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include bark spectral distortion (BSD), the perceptual speech quality measure (PSQM), modified BSD (MBSD), measuring normalizing blocks (MNB), PSQM+, the telecommunication objective speech quality assessment (TOSQA), the perceptual analysis measurement system (PAMS), and most recently the perceptual evaluation of speech quality (PESQ), which is specified by ITU-T recommendation P. 862 [7], as the international standard for testing networks and codecs. In the case of input-output based speech quality assessment, good correlations were observed, reaching up to 99 per cent in some cases [8].

The field of estimating the speech quality using only received speech without access to the input record is relatively new. Most recently, a couple of attempts to develop more credible non-intrusive speech quality measurements based on perceptual analysis have been reported. In 1994, Jin Liang and R. Kubichek [9] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results, in special cases achieving 90 per cent correlation.

Perceptually-based methods imitate the human listening method where many parameters and environmental effects are considered. R. Kubichek and Chiyi Jin [10, 11] used the vector quantization method, which yields up to 83 per cent correlation. Vector quantization has some disadvantages that yield poor results. One of these disadvantages is an inherent spectral distortion in representing the actual analysis vector. Since there is only a finite number of code book vectors, the process of choosing the "best" representation of a given spectral vector inherently is equivalent to quantizing the vector. It leads, by definition, to a certain level of quantization error. As the size of the code book increases, the quantization error decreases. However, with any finite code book there will always be some non-zero level of quantization error. Furthermore, the storage required for code book vectors is often significant. The larger we make the code book, the more storage space is required for the code book entries.

Another example of this is an output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal, reported in [12]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray *et al*. [13] reported a novel use of vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in a non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [14]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

In [15] a new output-based speech quality measure, which uses bark spectrum analysis and fifth order perceptual linear prediction PLP, was introduced. The measure is based on comparing the output speech to an artificial reference signal that is appropriately selected from optimally clustered reference code book, using the self-organizing map approach coupled with an enhanced k-means technique. The average correlation of this technique reached 0.85 for bark spectrum and 0.61 for PLP coefficient, respectively. Both short duration of speech records and limited number of speakers count as disadvantages in this study. Only two male speakers were used; one for training and the other one for testing.

In [16], Khalid Al-Mashouq and Mohammed Al-Shaye proposed a time-delay multilayer neural network model that can rate the speech quality after a proper learning stage. The learning set consists of features such as linear predictive coefficients (LPCs) and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached 0.87. However, the published result is not generalized due to the limited number of speech files used.

The latest research was introduced in 2004 [17]. This paper describes a new output-based approach that extends the original pseudo-reference framework by replacing the VQ code book with a discrete hidden Markov model. Correlation between objective and subjective scores shows good performance for text dependent and text independent cases of 0.88 and 0.92 respectively, but does not provide speaker independence. However, the only research in the mobile communication environment is by Al-Mashouq and Al-Shaye [18].

In this work we propose a new scheme for measuring speech quality over the network from the subscriber set. This method adopts a non-traditional approach and can be used as a relative index for assessing the quality of speech. Further, the method offers a significant advantage over traditional speech quality measurement as the network operator and subscriber can evaluate the speech quality for every call made under different conditions. This can be applied in a number of ways:

1. In present day multi-operator environments, service subscribers can easily use this method to choose between different networks.
2. Telecom regulatory bodies can apply this method to enact laws that will make operators charge less for less voice quality measured from their networks.
3. Network operators can use this method as a marketing tool and offer subscribers reduced tariffs whenever the speech quality of calls they make is lower than a particular threshold.

The proposed method is part of ongoing research that will involve investigation of metrics which are used to detect the call, obtaining the signal strength information, refreshing the value at every five milliseconds and finally implementing the application in a Symbian operating system.

2 The proposed method

Our ultimate goal in this work is to develop mobile handset software which will perform an automatic speech quality assessment for every telephone call. It should give an objective score for each call along with a periodic speech quality average. This, in turn, will give subscribers a practical way to assess the performance of different operators. Moreover, operators may want to give their employees a version of this software to do internal network auditing.

As a starting point, we collected a cellular speech database and its original version. Then, we built up a robust continuous speech recognizer. Finally, we derived an automatic machine score wherein correlation with its input-output counterpart is high. The speech databases used in most of the simulation stage are the well-known TIMIT and CTIMIT database set. CTIMIT is the cellular version of the TIMIT phonetic database. We chose these because of their large vocabulary and mixed speakers' accents. The hidden Markov models have to be trained using a clean speech database. The noisy speech database is used for testing. Each word of the database is segmented and phonemes assigned. Then that word is trained for the hidden Markov model parameters using the Baum-Welch algorithm.

For test word recognition, we used the forward or backward algorithm of the hidden Markov model to find the most likely word in the database. The recognized word's probability is categorized on 1-5 scale of quality. Then the test scores are correlated with the PESQ algorithm.

As stated earlier, our proposed method adapts a non-traditional approach and the aim is to use it as a relative index when assessing the quality of speech. This is applied in the following manner:

1. Upon a call being set up, the signal level values are taken every five milliseconds.
2. The criteria in Table 1 (below) are used to decide on the quality of the signal.
3. A score is then given to the sample collected every five milliseconds; with five being the best and 1 indicating very bad signal strength.
4. At the end of the call session a cumulative score is computed. Based on the score (ranging from 1 to 5) the speech quality is approximately computed.

Signal level range (dBm)	**Classification**
-120 to -95	Extremely bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very good

Table 1: Perceived quality rating of human speech

3. Emerging results

As stated earlier, our proposed method adopts a non-traditional approach and the aim is to use it as a relative index when assessing quality of speech. As a step towards the realization of our research object, the first and second versions of the software have been developed. These have met our set requirement of measuring signal strength every five milliseconds during a call and provide average strength of the signal at the end of the call. An excerpt of documentation of the first and second phase of the operating platform is shown in Appendix 1.

4. Summary

This report has examined a more liberal approach to speech quality measurement in telecommunication networks. It aims to equip the network subscriber with the opportunity to choose his or her telecom service provider based, among other key indices, upon speech quality. This will also afford service providers the capability of predicting customers' opinion of quality of service and the need for network optimization for continued customer satisfaction that will ensure loyalty and an increased subscriber base.

Appendix 1: Documentation of operating platform

2008/05/01 - 15:08:41 : : Current network info LocationAreaCode = 3341 CellId = 6826328
2008/05/01 - 15:08:41 : : Signal strength is = - 95 Db
2008/05/01 - 15:09:09 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/01 - 15:09:11 : : Current network info LocationAreaCode = 3341 CellId = 6827960
2008/05/01 - 15:09:11 : : Call drop observer -> Event : The current cell is changed
2008/05/01 - 15:09:11 : : Call drop observer -> Your call was dropped due to Hand-over
2008/05/01 - 15:09:15 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/01 - 15:09:15 : : Average signal strength is
2008/05/01 - 15:09:15 : : ***********************
2008/05/01 - 15:22:37 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/01 - 15:22:41 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/01 - 15:22:41 : : ***********************
2008/05/26 - 05:30:58 : : Current network info LocationAreaCode = 3341 CellId = 6827960
2008/05/26 - 05:30:58 : : Signal strength is = - 52 Db
2008/05/26 - 05:31:22 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/26 - 05:31:22 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/26 - 05:31:22 : : Average signal strength is - 52 Db (Very good)
2008/05/26 - 05:31:22 : : ***********************
2008/05/26 - 05:34:03 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/26 - 05:34:03 : : ***********************
2008/05/26 - 05:34:16 : : Current network info LocationAreaCode = 3341 CellId = 6827960
2008/05/26 - 05:34:16 : : Signal strength is = - 52 Db
2008/05/26 - 05:34:45 : : GPS coordinates lat = 24.7078 , long = 46.6619 , accuracy = 335.95 m, altitude = 612.0000 m , altitude accuracy = 109.50 m
2008/05/26 - 05:34:47 : : GPS coordinates lat = 24.7078 , long = 46.6614 , accuracy = 223.00 m, altitude = 612.0000 m , altitude accuracy = 100.00 m
2008/05/26 - 05:34:48 : : GPS coordinates lat = 24.7078 , long = 46.6616 , accuracy = 281.89 m, altitude = 612.0000 m , altitude accuracy = 89.50 m
2008/05/26 - 05:34:49 : : Call drop observer -> Event : Call state is changed. Phone status: Disconnecting
2008/05/26 - 05:34:50 : : Call drop observer -> Event : Call state is changed. Phone status: Idle
2008/05/26 - 05:34:50 : : Average signal strength is - 52 Db (Very good)
2008/05/26 - 05:34:50 : : ***********************
2008/05/26 - 05:34:50 : : GPS coordinates lat = 24.7077 , long = 46.6616 , accuracy = 315.13 m, altitude = 612.0000 m , altitude accuracy = 80.00 m
2008/05/26 - 05:34:51 : : GPS coordinates lat = 24.7078 , long = 46.6615 , accuracy = 290.06 m, altitude = 612.0000 m , altitude accuracy = 71.50

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