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RXQUAL *and* Voice Quality

RXQUAL and Voice Quality

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Introduction

It has only been a few years since GSM technology found its way from development laboratories into the field of commercial service. The complexity and multiple variations of all operational situations in the real world could never be simulated and tested in the laboratory, therefore many bugs and system configuration problems had to be discovered and resolved in the field. Consequently, the accepted "culture" of performing GSM field measurements slowly evolved, over a period of time, into a suite of test protocols using standard engineering and trouble shooting tools. This evolution still continues and is still being driven by the needs of technicians and engineers who are responsible for network implementation and operational stability.

Accordingly, the traditional and common approach used to investigate GSM radio link control procedures is with protocol analysers and first generation cellular verification systems. These engineering and trouble shooting tools extract diagnostic information from the GSM system, concentrating on parameters like RXQUAL, RXLEV and layer 3 signalling messages. This data is useful in network maintenance but is limited when addressing the important issue of Quality of Service (QoS) – particularly as viewed by the mobile phone user.

The subscriber's perception of "quality" – specifically voice quality – is still rarely taken into consideration. The primary reason being that radio technicians believe that a "human interpretation" of quality would not be a reliable, objective and reproducible measurement.

However, the subject of QoS – especially in cellular networks – is now being recognised as "critical to success" by more and more network operators. In today's competitive environment, it is important for GSM operators to be able to accurately assess performance and voice quality of their networks in order to maintain or extend market position.

From a subscriber's point of view, the relevant performance measure is the call success rate and received speech quality. These two very important QoS criteria have, however, been difficult to assess, quantify and analyse. The simpler approach, the RXQUAL or bit error rate (BER) method to assess speech quality has been preferred for ease of access and economic reasons. BER is readily available on the downlink – and it is commonly believed to correspond well to speech quality[1]. This idea – stemming from the limited agreement of BER and speech quality at very low and very high BER values – has generated some serious misunderstandings.

RXQUAL, unfortunately, does not represent "real life" conditions. This White Paper discusses inherent problems of RXQUAL and demonstrates that the measurement cannot be related to perceived speech quality in an accurate and reliable way.

QoS Criteria and Speech Quality

The subject of QoS in cellular networks is very broad and covers a wide variety of topics. For our purposes, it can best be defined by two groups of criteria: Call Procedure and Speech Quality.

Call procedure related QoS criteria:

Before call attempt:
- probability of no network service availability

If service available:
- probability of call attempt failure (uplink + downlink)

If call attempt successful:
- call set-up duration (uplink + downlink)
- probability for call drop

Speech quality related QoS criteria:

If call held successfully until desired release:
- actual perceived speech quality over call duration (uplink)
- actual perceived speech quality over call duration (downlink)
- statistical distribution of different quality levels (e.g. percentage of call duration suffering from lost syllables, echo, robot-voice, ping-pong)

It is commonly believed that call procedure and speech quality can be derived from engineering measurements. The call procedure related items would simply be taken from the signalling messages while the more subtle criteria of speech quality is derived from the downlink RXQUAL (this is also suggested by the name of this parameter).

Experienced engineers – responsible for quality assurance of cellular networks – know how difficult it is to categorise a call attempt into classes of call success by correctly interpreting signalling messages. The major problem in this regard is that the majority of available tools do not provide suitable post-processing capabilities. And, call success analysis is only the beginning of understanding QoS issues. **A more serious matter, arises when downlink RXQUAL is used to draw conclusions about speech quality. Results are always ambiguous and can be very misleading**. This White Paper will demonstrate the necessity of real speech quality measurement and explain in detail why downlink RXQUAL is not the suitable parameter for this purpose.

For a better understanding of the various points presented, some basic information about methods of speech quality measurement, speech transmission in GSM and the meaning of the RXQUAL parameter is provided.

Assessment of speech quality

It is extremely difficult to define a term as vague as speech quality; everybody has a different view of the subject. A pragmatic approach is as follows:

> If many people assess the "quality" of a particular speech sample, the average of all votes is taken as a measure of the speech quality of that sample.

This average is called the mean opinion score (MOS). In fact, it is a reasonable performance measure for GSM: as speech quality has a different meaning for every mobile phone user, the MOS (ideally) reflects the average perception of quality among the subscribers. ITU-T recommends 5 levels of quality:
"excellent" – "good" – "fair" – "poor" – "bad" and describes a procedural approach to determine MOS by listening tests under laboratory conditions. These tests had been applied in the selection process for the best suitable GSM speech codec.

Subjective versus measured speech quality

In practice – in an operational network – it is not feasible to assess speech quality at different locations by listening tests because of the enormous effort that such tests would require. Instead, one resorts to automated measuring methods that aim to approximate the MOS. There exists a wide variety of schemes, ranging from simply measuring the bit error rate to modelling the human hearing system. As a rule of thumb, more complex schemes yield more accurate results.

Any reasonable measurement method will agree well with speech quality if there are no transmission errors (quality must be good) and if there are many transmission errors (quality must be bad). However, the dividing line between acceptable and unacceptable speech quality lies somewhere in between. Therefore, it is important to employ a measurement method that meets these two requirements:

1. Yields accurate measurements in the intermediate quality range
2. Does not suffer from systematic errors.

The first requirement is illustrated in Figure 1. Scheme A yields accurate measurements for all levels of speech quality, whereas scheme B can deviate heavily from the target value at intermediate speech quality. Only scheme A is reliable around the level of "just acceptable" speech quality.

The second requirement is equally important. For example, a systematic error could result in always overestimating speech quality under specific circumstances; as a consequence, this would prevent detection of quality problems in that case.

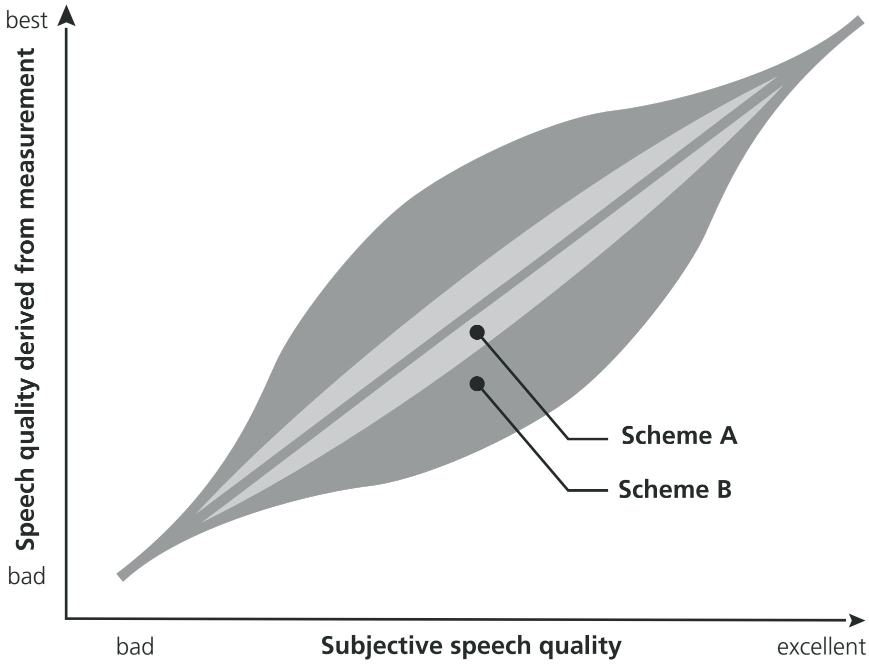

Figure1
Accurate (A) and inaccurate (B) measurement patterns of speech quality

Tests with the QVoice[3)] system demonstrate that speech quality measurement by automatic procedures provide sufficient accuracy to fully substitute listening tests in operational cellular networks or similar applications.

Overview of GSM Speech Transmission and Radio Channel Signal Processing

Speech transmission

The GSM speech transmission path is sketched in Figure 2. First, speech is compressed to (approximately) 20% of the input data. The resulting data stream is then protected by an error-control coding scheme. On the radio transmission path, various sources of errors can disturb the transmitted data. At the receiver, the channel decoder attempts to recover from these errors and delivers a "cleaned up" version of the received data. Finally, speech is reconstructed in the speech decompression block.

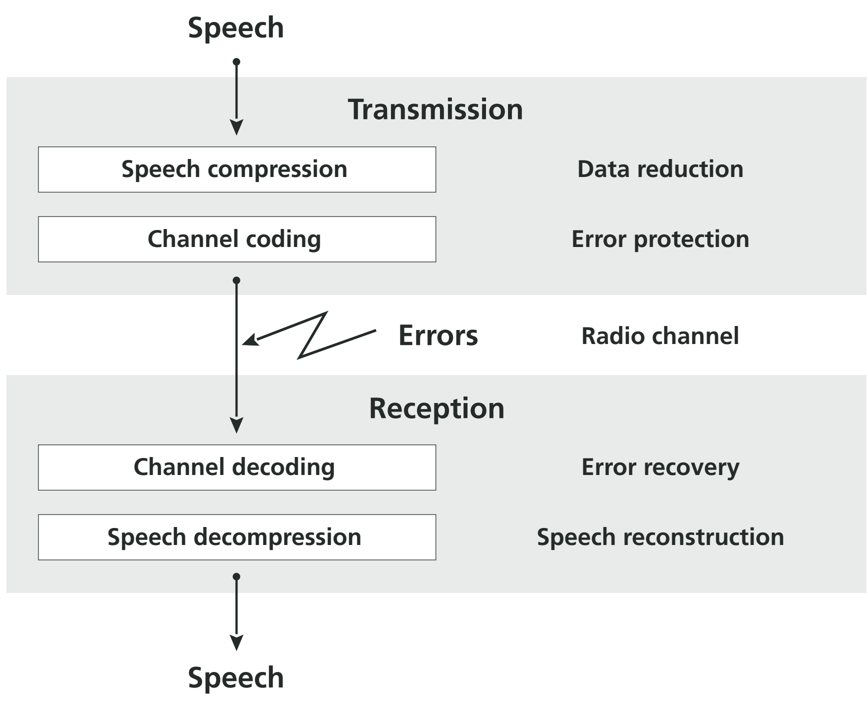

Figure 2:
GSM speech transmission

Bit error rate measurements in GSM

Two points are noteworthy. First, the speech compression algorithm is lossy. Consequently, some degradation of speech quality is unavoidable – even without any transmission errors. Second, no error-control scheme can cope with all possible error events. Bad transmission conditions usually result in unrecoverable errors, which, in turn, cause a degradation in speech quality. This White Paper considers only degradation effects resulting from transmission errors.

In GSM, bit error rate (BER) measurements are used for two purposes. On the one hand, they help to decide whether transmitter power should be changed (power control). On the other hand, BER assists in deciding whether a call should be attached to another base station (handover). In both cases, BER is only one among several criteria; moreover, moderately precise measurements suffice for both purposes.

A single BER measurement in GSM extends over a period of approximately 0.5 seconds and is reported as one of eight quality levels (RXQUAL_0...7). RXQUAL – estimated by backward coding of the decoded bit sequence and comparing it to the received bit sequence – is a measure of the raw bit error rate, and does not take into consideration channel coding.

Simulation system

Several of the arguments, particularly 1, 4 and 5 were proven by intensive simulation studies of the GSM transmission path[2)].

The actual system used included the following modules:

- Speech compression (GSM 06.10)
- Channel coding and interleaving (GSM 05.03)
- A simplified flat fading channel model
- Channel decoding and de-interleaving (GSM 05.03)
- Bad frame substitution (GSM 06.12)
- Speech decompression (GSM 06.10)

The speech samples used were produced under a wide variety of conditions.

Radio channel signal processing

For better understanding the following arguments, the most important steps of the radio signal processing and transmission sequence has been outlined in Table A. It is not intended to be a complete listing but to show the "influential" segments in terms of QoS discussions. Table A is the backbone to several of the arguments and will be referred to throughout the White Paper.

		Processing Entity	Transmitted Information	Remarks
Transmission	**1**	Human speaker or QVoice	Original voice	Start of signal processing QVoice reference sample
	2	Transmission system and switches (ISDN and PSTN)	Voice coded in 64 Kbit/s A-law	Potential source of echo and distortion
	3	Voice activity detection and discontinous transmission	Voice coded in 64 Kbit/s A-law or Silence indication	Potential source of voice clipping
	4	Speech coder	Speech frame: 20 ms duration, 260 bits 182 class 1 bits 78 class 2 bits	
	5	Convolutional coding (only for class 1 and 3 parity bits)	Protected speech frame: 20 ms duration, 456 bits 378 class 1 bits (protected) 78 class 2 bits (unprotected)	
	6	Interleaving	Partitioning of speech frame: 8 subsequent bits are spread into 8 different time-slots (each 57 bits)	
	7	Burst formatter	2 x 57 data bits 26 bit training sequence guard bits	
	8	GMSK modulator	Original GMSK modulated baseband signal	
	9	RF mixer - fixed carrier frequency - frequency hopping	Original GMSK modulated radio signal	

Table A:
Processing sequence of a voice signal trough a GSM network

		Processing Entity	Transmitted Information	Remarks
Radio Channel	**10**	Radio channel - path loss - slow lognormal fading - multipath propagation - fast rayleigh fading - intersymbol interference - noise - interference - doppler shift	Distorted GMSK modulated Radio channel	
Reception	**11**	Receiver front end - antenna diversity reception - single carrier of FH	Distorted GMSK modulated Basband signal	
	12	Receiver: - adaptive matched filter and equaliser	8 received bursts: 26 bit distorted training sequence 2 x 57 distorted data bits	Benefiting from antenna and path diversity
	13	De-interleaver	Received protected speech frame: 20 ms duration, 456 bits 378 class 1 bits (protected) 78 class 2 bits (unprotected)	
	14	Channel decoder error correction (only on 378 class 1 bits)	Received speech frame: 20 ms duration, 260 bits 182 class 1 bits 78 class 2 bits CRC parity check result	RXQUAL estimation different efficiency of error correction depending on speed, fading pattern, frequency hopping
	15	If bad frame indication based on CRC test or stealing flag for FACCH use or otherwise invalid voice data, speech frame substituion else if silence indication flag comfort noise generation else speech decoder	Reconstructed voice in 64 Kbit/s	BFI may fail with a 12.5% probability due to 3 bit CRC check creating ping-pong sound
	16	Human ear or QVoice[3)]		Evaluator of speech quality

Argument No. 1: RXQUAL Does Not Consider the Varying Efficiency of Interleaving and Bit Error Correction Under Different Environmental Conditions.

The benefits of interleaving and channel coding

As shown in step 4 of Table A, the speech coder converts segments of 20 ms of speech into 182 class 1 bits and 78 class 2 bits – a total of 260 bits. Class 1 bits are more important for reliable coding of speech. Therefore, only class 1 bits are protected by 2:1 convolutional coding. See step 5 of Table A. Class 2 bits are left unprotected because they have only minor significance for speech quality. This yields 456 coded bits for every 20 ms of speech. Within the interleaver – step 6 of Table A – subsequent bits are separated so that they are transmitted within 8 different time-slots. This calculates to be up to 40 ms in delayed transmission.

This procedure is necessary in order to handle short term Rayleigh fading on the radio channel. This is step 10 of Table A. The process, however, leaves some interesting effects demanding careful consideration, such as shown in the following examples which compare the efficiency of interleaving in combination with error correction coding of class 1 bits for different speeds of the mobile phone.

Fast mobile phone

Figure 3 shows a typical stationary Rayleigh fading pattern that is caused by multi path propagation such as in urban areas.

If a signal breakdown lasts 10 ms then every 4th bit of the original non-interleaved bit sequence may be damaged. This can be repaired efficiently and effectively by the 2:1 redundancy of the channel coding taking place in step 14. Assuming that a deep fading notch has a width of $\lambda/5 = 6$ cm (being the wavelength of the 900 MHz RF signal), then a speed of 120 km/h would be sufficient to reduce the time of fading to 1.8 ms. This means that the speech reference although damaged during a signal breakdown would arrive in good condition at the receiving end, step 16 of Table A.

Slow mobile phone

A more critical situation arises when the mobile phone is stationary or moving very slowly such as pedestrian speed, approximately 7 km/h. In this case deep fading of 6 cm width would last 30 ms – or in other terms nearly the full interleaving width. Under these conditions, GSM signal processing would generate an uninterrupted sequence of damaged bits and the error correction taking place in step 14 of Table A would fail. Therefore, class 1 bits would suffer from fading and the transmitted speech would arrive in a degraded state at the receiving end, step 16 of Table A.

The crucial question with respect to the theme of this White Paper is: Does RXQUAL report different speech qualities for the two situations described above? The answer is no.

RXQUAL is completely insensitive to the differences of these realistic situations of fast and slow moving mobile phones. This is because the measurement is only an estimate of the raw bit error ratio and therefore complete ignores the benefits of channel coding and interleaving as depicted in steps 5, 6, 13 and 14 of Table A. RXQUAL is the estimate for the BER of unprotected class 2 bits rather than for class 1 bits.

Test Results

The effect of the Rayleigh fading channel is illustrated in Figure 3 and Figure 4 for the Rayleigh fading channel. An English speech sample was transmitted (by simulation) to a slow mobile (7.2 km/h, 900 MHz) and to a fast-moving mobile (120 km/h, 900 MHz).

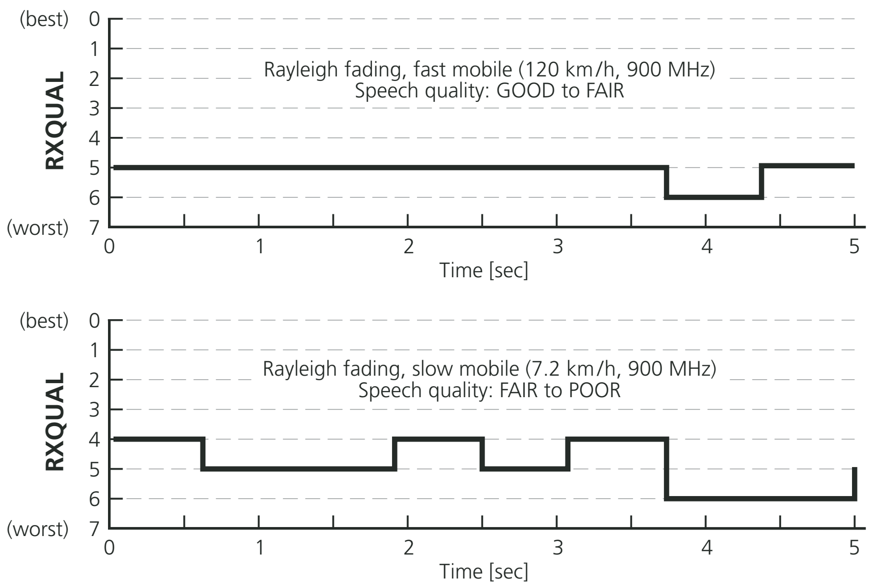

Figure 3:
RXQUAL vs. time [s] for Rayleigh fading for two mobiles of different speed. Note that RXQUAL of the slow mobile is at least as good most of time.

Sample of fast moving mobile typical of a vehicle installed phone

Sample of slow moving mobile typical of a hand-held pedestrian phone

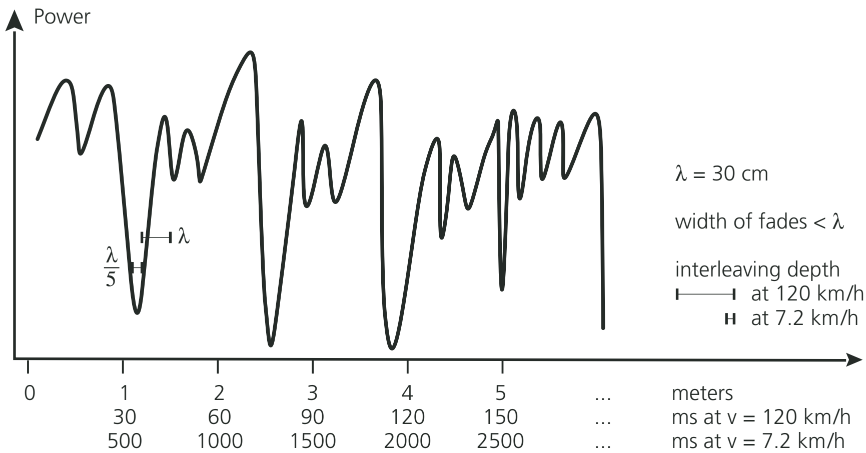

Figure 4:
Stationary fading pattern

Fades last much longer for the slow mobile, thus resulting in much longer error bursts. In spite of that, the raw BER measurements described by RXQUAL are very similar due to their averaging effect; the slow mobile RXQUAL was as good as or better than that of the fast mobile most of the time. Nevertheless, speech quality was clearly better for the fast mobile.

The conclusion from this analysis is that RXQUAL measurements can not take into account the speed of a mobile phone that has a significant impact on speech quality.

The benefit of frequency hopping

Frequency hopping is the GSM technology to combat Rayleigh fading and to mitigate co-channel and adjacent channel interference.

Fading environment

Frequency hopping would be effective when applied within the fading scenario described above for a slow moving mobile phone. The reason for this is that frequencies which are sufficiently spaced do not generate correlated fading patterns. Figure 4 shows a typical fading pattern. This means that statistical characteristics – like average field strength and field strength probability distribution – would be identical while the individual location of fading is not correlated. Switching, therefore, from one frequency to another shifts fading notches when mobile phones are slowly moving or stationary. The result for signal processing efficiency would then be equivalent to the fast moving mobile example presented earlier.

Figure 5 shows BER (class 1) for a slow moving mobile – with 5 km/h for 4 different cases:

a) without error correction (class 2, RXQUAL), FH applied

b) without error correction (class 2, RXQUAL), no FH

c) with error correction (class 1), FH applied

d) with error correction (class 1), no FH

Cases c) and d) of Figure 5 show that frequency hopping reduces class 1 BER by a factor of more than 10. Cases a) and b) of Figure 5, in comparison, cannot be separated (same curve).

One can conclude that RXQUAL cannot distinguish the obvious benefits of frequency hopping when considering voice quality in fading environments.

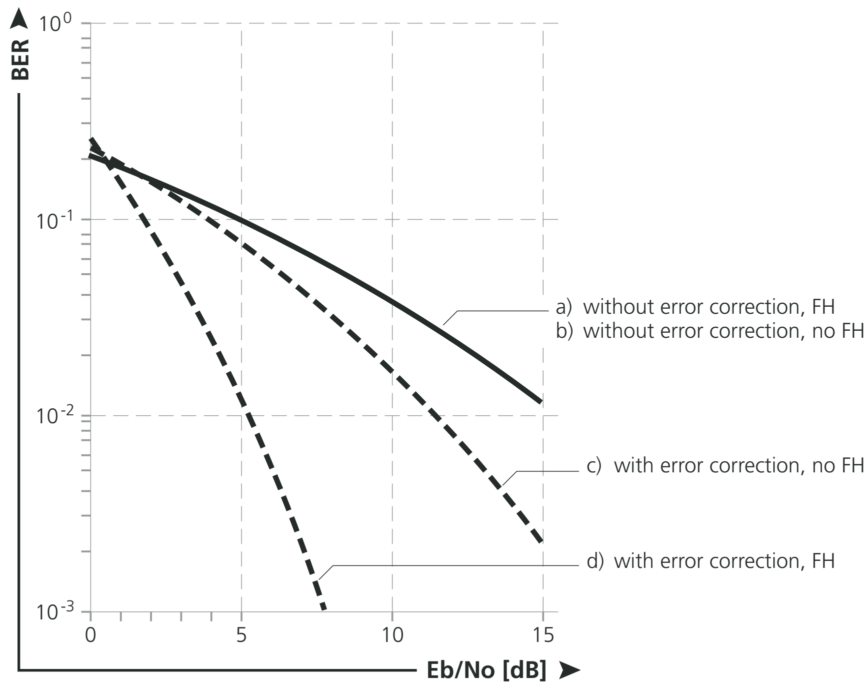

Figure 5:
Depending of BER (class) on Frequency Hopping and Error Correction Coding

flat fading without diversity
v = 5 km/h

Source: Lucent Technologies
Nürnberg, Germany

Argument No. 2: RXQUAL Does Not Consider Quality Degradation Caused by Stolen Speech Frames

Stealing speech frames is a common procedure in GSM when fast signalling is required during the call period.

Normally, the traffic channel for full rate speech (TCH/FR) is accompanied by a slow associated control channel (SACCH) which is used to transfer non-time critical information. SACCH has a recurrence interval of 480 ms and an interleaving delay of 360 ms. It is used to transfer system information messages in a regular way in downlink or to transfer MS measurement result messages in uplink.

However, if handover is required and executed, an immediate response is necessary and therefore the traffic channel capacity is used. This is achieved by marking a speech frame of 20 ms duration as stolen (i.e. invalid for speech) and inserting signalling messages instead. The flag is called Stealing Flag and the stolen TCH becomes FACCH (fast associated control channel).

When a speech frame is lost in a controlled way (setting the stealing flag), it is replaced by another type of speech data – mostly a repetition of the last valid speech frame, which is muted more and more for subsequent stolen frames. This occurs at step 15 of Table A).

Fast signalling is a required standard event during a call at each handover execution. Interruption of the valid speech transmission for a successful handover is about 100 to 200 ms, actual time being dependent mostly on how well the old and new cells are synchronised. **Handover failures could however cause significant speech interruptions and should be monitored.** The actual degradation due to stolen frames and the efficiency of speech frame substitution can be monitored with advanced quality testing and analysis systems such as QVoice[3)].

RXQUAL is not a suitable test parameter in determining quality effects of erased frames. It does not consider stolen speech frames nor resulting speech quality degradation in any way.

Argument No. 3: RXQUAL Does Not Detect Echo or Other PSTN Quality Impairing Effects

BER measurements only take into account errors on the radio transmission path (Figure 2). But the transmitted bits have no meaning to the measurement unit – they are just bits. Thus, distortions prior to the speech compression unit remain undetected by BER measurements. This includes echo, crosstalk and noise in the analogue part of the network.

Echoes originate within the analogue part of a network during speech transmission. A fixed network telephone, for example, might have a design flaw or construction error that allows an acoustical feedback loop to form from the receiver back into the transmitting end.

Similar effects occur in the electrical circuitry of Public Switched Telephone Network (PSTN) switches or analogue transmission devices. Within pure analogue networks these effects do not cause echo, in the sense of human perception, because the signals are travelling over wires or airways without built-in delays like buffering or coding. The resulting time delay between the main signal and the echo is too short for human perception.

When a PSTN is connected to a digital GSM network the situation is different. GSM has several processing steps where signals are buffered and transmitted with certain time delays. Speech coding which is taking place in step 4 of Table A and interleaving, at step 6 of Table A causes a total one way delay of 38 ms. That means it takes 38 ms after a message is spoken to reach the PSTN network. The path of the acoustical echo and the way back into the GSM mobile phone – steps 13 and 15 of Table A – will add up to another delay of at least 38 ms. Delays of this order of magnitude are perceived as annoying echo and turn out to be very irritating to the speaker as well as the listener on the receiving end of the link.

As an example, consider the case where a speech signal is severely impaired by crosstalk in the wired part of the network. Referring to Figure 2, the distorted speech signal would be compressed, error protection added, and the resulting bit stream transmitted. At the receiving end, BER measurements would only detect errors on the radio link (i.e. in the bit stream), but not distortions of the input speech signal. The speech signal at the output side of Figure 2 would remain impaired by crosstalk (poor speech quality), even if the radio link transmission was perfect – and thus RXQUAL also measuring perfect.

The conclusion is that RXQUAL cannot detect echo or other disturbing PSTN effects. It is important, however, for an operator to continuously monitor the network for these effects before a high voice quality can be assured.

Argument No. 4: Speech Quality is the Only Parameter to Detect Defects Within Voice Processing Circuits

Argument No 3 discussed several effects of the PSTN network which might degrade speech quality and which are not detectable by surveying the radio path which is provided by the RXQUAL parameter. Under certain circumstances the non-radio related GSM network infrastructure may also turn out to be a source of speech quality degradation and even outage of the speech in uplink, downlink or both. The following functional units have to be considered:

- Voice activity detection and discontinuous transmission (step 3 of Table A)
- Voice coding done by TRAU (transcoder/rate adaptation unit) (step 4 of Table A)
- Voice decoding done by TRAU (step 15 of Table A)

The TRAU is placed between BSC and MSC and therefore not within the scope of RXQUAL measurements. **In case of malfunction one of the processor boards, assigned to a given traffic channel, the speech quality, can be heavily degraded. This may happen without notice of the operation and maintenance centre because self-testing capabilities of individual boards are not always implemented.**

Argument No. 5: RXQUAL Interpretation Suffers from Coarse Quantisation

In GSM, the BER measurements are quantified to eight RXQUAL levels. The coarse quantification makes it possible to have different speech qualities with different BERs that are quantified to the same RXQUAL level. This means that it would be impossible to distinguish different levels of speech quality from looking at just RXQUAL measurements.

An example is shown in Figure 6, in which an English speech sample was transmitted (by simulation) over two AWGN (Additive White Gaussian Noise) channels with different Bit Error Rates (different BERs can arise for several reasons such as different distances of the mobiles to the base station). As both BER values belong to the same RXQUAL level – the result of the coarse quantification algorithm – the RXQUAL measurements are identical. In spite of that, speech quality on the channel with the lower BER was clearly better. **Therefore, coarse RXQUAL quantification may yield ambiguities in speech quality.**

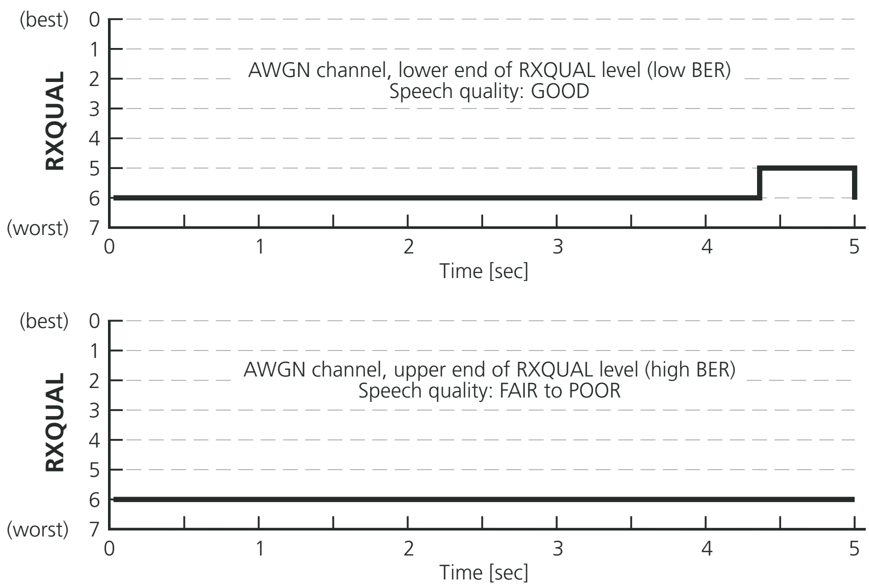

Figure 6:
RXQUAL vs time(s) for AWGN with different BER, but identical RXQUAL

Argument No. 6: RXQUAL Does Not Recognise Whether Bad Speech Frame Indication Failed

Every GSM user has been annoyed by the characteristic sound which resembles a bouncing Ping-Pong ball. Sometimes, this noise is quite useful because it is an early announcement before loosing the connection. However, the designers of GSM did not really plan to include such noise in the system. The "Ping-Pong" sound is caused by bad speech frames which the GSM system itself is unable to detect.

The designers of GSM included a 3 bit cyclic redundancy (parity) check at step 14 in Table A. The design goal was to recognise bad speech frames in order to avoid the creation of sounds which are generated when feeding the speech decoder with spurious bit sequences. The recognition is only based on a 3 bit CRC check; and, if the test reports a failure the Bad Frame Indication flag is set. As a consequence of a BFI flag set, a substitution is made, replacing the bad frame with the last valid frame. In cases of multiple subsequent BFI the acoustical signal is muted gradually to suppress annoying or strange sounds. After a valid frame is detected muting is cancelled.

Because only 3 bits have been reserved for BFI the probability for a non-detected bad frame and the potential creation of "dropping bottle" sound is 12.5%.

RXQUAL is not suitable to distinguish "Ping-Pong" sound from moderate degradation of speech quality caused by frame substitution. By contrast, state-of-the-art quality testing and analysis systems can detect this and other annoying effects. Being able to accurately detect these quality degrading effects is of particular importance because future half rate speech coders – or improved full rate coders – will most probable be more stable against erroneous sound – and a poor RXQUAL does not necessarily imply presence of artificial sounds.

Argument No. 7: RXQUAL Does Not Correctly Describe Speech Quality after Activating Frequency Hopping

Operators who decide to operate in switch on mode of frequency hopping are seeking improved speech quality for following purposes:

- To combat fading caused by Rayleigh fading for slow moving mobiles
- Allow mitigation of co-channel or adjacent channel interference for creation of additional capacity

The first item was treated extensively in Argument 1 The second is treated herein:

Background on frequency planning

The frequency spectrum is an extremely valuable resource. One fact derived from the US auctions was that for every US citizen under radio coverage, a license for mobile operation carried a value of 20 to 50 US dollars. Network planning engineers, not surprisingly, seek to optimise this resource by reusing frequencies within short distances in order to obtain the maximum number of frequency channels per cell. This leads to a non-diminishing probability of mutual interference P_{int}.

The P_{int} function states that for a given location and time – within a service area of a cell – the ratio of the wanted signal strength (C) to the sum (I) of all interfering signals from the same or adjacent channels is lower than the system threshold.

In GSM, the system threshold for a "good" and "realistic practical" quality is defined as the following:

$(C/I)_{thr} = 9$ dB

The planning goal for a network is usually in the order of:

$P_{int} = 1\%$

During peak traffic periods, a network with lower levels of interference can only be realised with exponentially increasing costs.

Interference on individual channels

Frequencies are specifically assigned to different cells according to traffic requirements and not according to fixed frequency groups of equal size. The reason being is that cellular traffic density – as measured in Erlang/km^2 can vary greatly between urban and rural areas – and the number of TRX per cell has a typical range between 1 and 6. A consequence of this – "allocation by cell traffic need" – is that each frequency in a given cell may find its co-channel and adjacent channel interferers in different cells with individual propagation path losses.

Furthermore, the actual activity and strength of individual interferers is dependent on random variables indicating the following:

- Whether the interfering TRX (on the relevant time-slots) has a traffic channel TCH assigned
- In case of TCH being assigned – whether it is currently in an active time of discontinuous transmission (DTX)
- Which level of power control is actually applied

These complex conditions lead to individual interference situations for each TRX at a specific time and location. By way of a case study, Table B depicts a typical instantaneous interference situation in a cell with 4 TRX at a random time, time-slot and location:

Table B:
Case study
Instantaneous interference situation
in case of Hopping/No-Hopping

TRX No	C/I	RXQUAL case of no FH	Speech qual. case of no FH	RXQUAL case of FH	Speech qual. case of FH
1	30 dB	0	excellent	5	good
2	11 dB	2	good	5	good
3	5 dB	7	bad	5	good
4	13 dB	1	excellent	5	good

Table B shows a typical situation of various C/I values over different frequencies. In the "no FH case" TRX 3 – with highest interference load of C/I = 5 dB – turns out to be unusable. The probability of handover failure, and subsequent dropped calls would increase dramatically. Call attempts would also result in many connect failures on TRX 3. The effective usable capacity of the cell would be only 75% of the installed capacity.

Frequency hopping creates interferer diversity

Collisions on single traffic channels with co-channel and adjacent channel frequencies that lead to intolerable interference – under some circumstances – are unavoidable. Frequency hopping was included in the GSM design to smooth out these interference collisions by creating Interferer Diversity. The process also allows for the repair of spurious bits – on a single channel – by using the good transmission characteristics of the other channels. Because the GSM signal processing was designed with a high level of efficiency, good reception quality is possible even if one of the four frequencies of a hopping sequence is completely blocked (resulting in a bit error rate for that frequency of 50%).

Frequency hopping is a key feature in economic operation of a network because frequency plans no longer have to be designed for worst case (worst frequency) scenarios. With frequency hopping, the targeted QoS can be reached using a smoothed average scenario. The potential gain is an increase of traffic capacity with revenues per cell from 10% up to 30%. Frequency hopping makes colliding frequencies again usable. Discontinuous Transmission (DTX), Power Control and temporarily unused channels all contribute to a reduced interference level. This becomes usable "space" to frequency hopping and as a consequence QoS or capacity can be improved.

Measuring the benefits of frequency hopping – case study

After activation of frequency hopping, one GSM network operator reported the ambiguous experience of QoS improvement on the one hand with indications of degradation by RXQUAL on the other hand! Described below are the observations followed by the author's explanation.

QoS improvements: Measurements of call attempt success rates and handover success rates showed some improvement while the number of dropped calls decreased (in a medium loaded network, frequency hopping may reduce the number of dropped calls about 20% of the previous rate).

Indications for QoS degradation: Contradictory to the observed quality improvements, the number of "handovers caused by bad quality" had increased steeply. In fact, statistical measurements of RXQUAL reported an average increase of about one unit. Under normal circumstances this means a serious degradation of transmission quality.

Interpretation of the results: The RXQUAL parameter does not increase linearly with the error rate of unprotected bits but with its logarithm. (RXQUAL increases by one unit if the bit error rate is doubled or decreases by 2 units if the bit error rate is divided by 4). This explains the RXQUAL values shown in Table B above.

In frequency hopping the bit error rates (for unprotected bits) for the different hopping sequence frequencies are averaged – and mapped into a $RXQUAL_{FH}$ value for the hopping channel. This $RXQUAL_{FH}$ value is not calculated as the arithmetical average of the $RXQUAL_{TRXi}$ values of the individual transceivers TRX_i in the non hopping mode:

$$RXQUAL_{FH} \geq \text{nint}\,[\Sigma_{i=1,n}\ RXQUAL_{TRXi} / n]$$

where n is the number of TRXs and nint [ ] is the nearest integer function. Examples are given in the Table C.

Table C:
Examples of RXQUAL calculations for hopping channels

$RXQUAL_{FH}$	$RXQUAL_{TRXi}$ i=1, n	average $RXQUAL_{TRXi}$	potential speech quality
5	0, 2, 7, 1	2.50	good
5	5, 5, 5, 5	5.00	fair
4	0, 6, 0, 0	1.50	excellent
2	0, 0, 4, 1	1.25	excellent
1	3, 0, 0, 0	0.75	excellent
1	0, 0, 0, 2	0.50	excellent
0	0, 1, 0, 0	0.25	excellent

The potential speech quality is included in the table, but there is not a unique mapping or correlation to RXQUAL. The reason being that speech quality is closer related to the protected class 1 bits, than to the unprotected class 2 bits – which are related to RXQUAL.

$RXQUAL_{FH}$ is a very questionable parameter as a quality figure for two main reasons:

- **The same value of $RXQUAL_{FH}$ can yield significantly different speech qualities as depicted in rows 1 and 2 of Table C**
- **$RXQUAL_{FH}$ has a different mapping to speech quality than $RXQUAL_{noFH}$. This is caused by the dominance of bad frequencies in the $RXQUAL_{FH}$ calculation and by the correction of these errors in the GSM signal processing.**

Quality testing system in BSS parameter tuning

Even when frequency hopping does not have a direct effect on class 2 bit error rates, a degradation of the average RXQUAL is reported as an artefact in the calculation. This explains the case for increased rate of "handovers due to bad quality".

On the other hand, the average resulting speech quality along with other QoS parameters in the network can be demonstrated to have improved using sophisticated quality testing equipment. Consequently, parameter threshold settings for power control and handover algorithms must be modified upon switch-on of frequency hopping.

It can be seen that RXQUAL values, again, give only a vague and ambiguities picture for assessing FH benefits in interference mitigation. Furthermore, speech quality measurements are required to find suitable RXQUAL threshold settings in power control and handover algorithms after introduction of frequency hopping.

The last chapter of this White Paper presents QVoice[3)]. Dedicated to quality test and analysis, the QVoice system can measure resulting improvements of speech quality. Furthermore, the system helps in setting handover and power control parameters, because it provides stable and reproducible criteria. Without relying on objective measurements, there is a danger of misinterpretation of observations and incorrect tuning.

Argument No. 8: In Case of EHR Speech Codec Introduction RXQUAL$_{EFR}$ Would Have a Different Mapping of Speech Quality

The initial goal for the introduction of enhanced full rate coders is to improve speech quality. This should be possible by use of higher processing power expected from the latest generation of GSM chips.

The RXQUAL parameter – as the measure of unprotected bit error rate will survive the introduction of enhanced full rate. However, the different levels of RXQUAL and corresponding speech quality for EFR codecs will show significant differences from what is known from full rate codec. **Therefore, the interpretation of RXQUAL measurements has to be revised after introduction of EFR, by means of independent voice quality measurements.**

Argument No. 9: Speech Quality is the Only Parameter to Compare Networks of Different Technologies

RXQUAL has been specifically defined in combination with the full rate speech coding according by GSM recommendations. Therefore, RXQUAL might be a helpful supplementary parameter in addition to speech quality in comparing the quality of GSM900, DCS1800 and PCS1900 networks with one another. This is possible because the algorithm of speech coding is identical.

Significant constraints in using RXQUAL for comparisons have to be considered following the introduction of half rate speech coding as mentioned under Argument No. 8; or, implementation of improved full rate speech coding which is currently under consideration within the GSM community. RXQUAL is still available but it has a different mapping to quality.

For competitive analysis between completely different technologies, QoS parameters as shown in the introduction of this White Paper are the only way. In the US market, there is for example, CDMA with variable rate voice coder up to 13 kbit/s, TDMA with 6.5 kbit/s, PCS at 1900 (=GSM) with a different 13 kbit/s coder, AMPS with 30 kHz FM and NAMPS with 10 kHz FM. These will all be in strong technological competition against each other. **Only the subscriber's ear or voice quality analysis systems can establish a ranking between different cellular and PCS technologies. RXQUAL will not find a corresponding parameter in other technologies with the same meaning to support pan-technology comparisons.**

Argument No. 10: Downlink RXQUAL May Fail in Detecting QoS Degradation in Uplink Direction

The asymmetry of uplink and downlink quality may have different causes summarised below:

System related causes:

- Unbalanced power budget
- Different equalisation depths within base station and mobile phone

Causes related to interference on the radio channel:

- Different levels of radio activity up/down
- Inter-modulation or receiver blocking caused by 3rd party interference
- Different topology of co-channel and adjacent channel interference for uplink and downlink
- Near/far effect

Propagation characteristics:

- Different location of Rayleigh fades

Most of the asymmetric effects indicate severe malfunction. Therefore, the monitoring of uplink and downlink with the same method is recommended. Failure to do so may leave QoS degradation undetected.

System related causes unbalanced power budget

The Theorem of Reciprocity tells that the path loss over the physical propagation environment is the same for uplink and downlink direction. Therefore, the transmission and reception characteristics of the GSM system shall allow for the same path loss as well. The small difference in frequency of about 5% is negligible below 1 dB. The two system paths are as follows:

The path loss is defined as

$$L := EIRP - P_{rec}$$

where

EIRP	Effective Isotropic Radiated Power
P_{rec}	Received power with (hypothetical) isotropic antenna

The system gain is as $EIRP_{max}$ - P_{recmin}:

for downlink:

$EIRP_{BTSmax}$ = maximum BTS power
– combiner and filter loss
– antenna cable loss
+ antenna gain

$P_{MSrecmin}$ = GSM reference sensitivity for mobile phone class i
+ antenna cable loss (if existing)
– antenna gain

for uplink:

$EIRP_{MSmax}$ = maximum transmission power of MS class i
– antenna cable loss (if existent)
+ antenna gain

$P_{BTSrecmin}$ = GSM reference sensitivity for BTS
+ gain of tower mounted antenna preamplifier (if existing)
+ antenna diversity gain (if existing)
+ antenna cable loss
– antenna gain

A balanced power budget is achieved if and only if

$$system\_gain_{up} = system\_gain_{down}$$

which is equivalent to

$$EIRP_{BTSmax} - P_{MSrecmin} = EIRP_{MSmax} - P_{BTSrecmin}$$

In reality, this relationship is only achieved with an accuracy of ±5 dB, depending primarily on the employed mobile phone class and the initial network design goals. Therefore, uplink and downlink may have different power ranges which may lead to significant performance differences on the cell periphery or at indoor locations.

Downlink measurements of RXQUAL may not be used to draw conclusions about uplink quality. Major quality differences between uplink and downlink might be an indication for BTS or antenna malfunction.

Different equalisation depths within base station and mobile phone

Another minor difference between uplink and downlink might appear in propagation environments with long multipath delays. The GSM specification is to equalise delays of up to 16 µs (corresponding to 4 bits duration) – technically, more is possible, based on the length of the training sequence which is 26 bits.

Mobile phone manufacturers will be forced by price competition to stay with the minimum specification and because multipath delay capability is not a consumer feature. But base station manufacturers might invest in more processing capability to achieve longer delay compensation up to perhaps 22 µs. This could also contribute to different quality in uplink and downlink direction.

Causes related to interference on the radio channel

Co-channel and adjacent channel interference depends on the activity of frequencies in the network since levels of radio activity in uplink and downlink are, in general, different. This is summarised in Table D.

Different levels of radio transmission activity

Table D: Factors with Impact on radio transmission activity

	MS to BTS	BTS to MS
Duty cycle on BCCH frequency	$P_{active} = T_{load} / N_{TCH}$	100%
Duty cycle on non-BCCH frequency	$P_{active} = T_{load} / N_{TCH}$	$P_{active} = T_{load} / N_{TCH}$
Power control	mandatory to cope with near/far effect	not for BCCH-freq. optional for non-BCCH-freq.
DTX enabled	mostly used because of longer talk time	not for BCCH-freq. optional for non-BCCH-freq.

The probability that a traffic channel is activated is described by the following equation:

$$P_{active} = T_{load} / N_{TCH}$$

The average traffic load (T_{load}) of the radio cell is given in Erlang (the average number of calls done at the same time within the cell). T_{load} is heavily dependent on time of day and on weekly and seasonal variations. N_{TCH} is the number of traffic channels configured within the cell.

After launch of a new network P_{active} will be close to zero due to the small number of subscribers. Therefore, interference within young networks can be dominated by downlink interference on BCCH frequencies. However, if the frequency assignment was not restricted to select BCCH frequencies – from a small subset of the total spectrum – that type of interference (in young networks) would be negligible.

But even in a mature network P_{active} will rarely exceed 60% to 75%. This is caused by the requirement that blocking probabilities (probability to find only busy channels at call set-up or incoming handover) have been specified by most operators to figures of 0.5% to 5%. The number of traffic channels N_{TCH} must exceed the average traffic in order to match the blocking requirement. The relation is given in Table E.

Table E:
Relationship of traffic channels to average traffic

N_{TRX}	N_{TRX}	T_{loadi}(2% blk)	P_{active}
1	7	2.9 erl	< 0.42
2	14	8.2 erl	< 0.59
3	22	14.9 erl	< 0.69
4	29	21.0 erl	< 0.73

Mature networks also have an unbalanced load between the uplink and downlink direction. This unbalance can be even more biased due to different DTX- and power control configurations in uplink and downlink.

Intermodulation or receiver blocking conditions

When designing a GSM network, not only the self-made co-channel and adjacent channel interference should be considered as potential sources of degradation. There are also other systems operating in neighbouring bands of the spectrum that should be taken into consideration. The mechanisms of degradation are many fold.

Third party unauthorised use of the allocated GSM band

In a number of countries the supervision of the spectrum by a state's authority is underdeveloped and unauthorised users of the spectrum are not identified by regular spectrum surveillance and prosecuted. As a consequence a GSM network could suffer from 3rd party interference. Interference could occur in uplink and downlink direction. Downlink interference is mostly limited to a certain area in the environment of the interfering transmitter, while uplink interference effectively reduces the radius of the cell.

Surveys using spectrum analysers connected to the base station antennas before network launch are strongly recommendable. In the running network, QVoice best in combination with OMC performance can provide hints about location and frequency of a potential interference. Further measurements should be done by use of high sensitivity spectrum analysers.

Spurious emissions

Sometimes, sites are shared by the GSM operator and other users. In particular, PTTs who own an old analogue network like to use the same sites when introducing GSM. Operating frequencies of the co-sited equipment will be different. However, the attenuation curves from the transmitters have a limited steepness and therefore, some power is radiated in certain distance from the transmitter frequency. A GSM base station receiver's performance is degraded if interfering power density of -114 dBm/200 kHz is exceeded. A typical transmitter of an analogue AMPS or TACS network has a power density of 40 dBm/30 kHz. The attenuation required for operation is 162 dB!

Normally, antenna decoupling accounts for about 50 dB. For large band separation of several 10 MHz the transmitter power density has decreased sufficiently. A very critical case is given when AMPS systems are sharing sites with GSM networks:

AMPS	uplink:	825 - 845 MHz	downlink:	870 - 890 MHz
GSM	uplink:	890 - 915 MHz	downlink:	935 - 960 MHz

The strong AMPS downlink is immediately adjacent to the GSM uplink (e.g. used in Taiwan). This leads to the necessity that a guard band has to be reserved. This is a waste of spectrum resources, and therefore, the guard band will be so small that a certain degradation by spurious emissions might not be excluded. These degradation will become obvious as a worse quality in uplink direction which is immediately measurable.

Receiver blocking

Another undesired characteristics of a radio receiver is the so-called blocking. Blocking is a desensitisation of the receiver for the wanted signal. It occurs if a strong signal arrives at the receiver even if it has zero power density at the frequency of the wanted signal. The blocking specification, for example, for a GSM receiver (i.e. less than 3 dB desensitisation) for an out-of-band signal is 0 dBm according to GSM recommendation 05.07..

Receiver blocking can be a problem in all cases if a strong transmitter is present. At the base station site it can be removed by additional filter equipment. This is not possible if mobile phones are affected (see also next paragraph “Intermodulation”).

Intermodulation

Intermodulation is another effect by which interfering signals within the used GSM band can be created from adjacent foreign signals out of that band. If strong foreign signals at frequencies f^1 and f^2 are arriving at a receiver other signals at frequencies $2f^1 - f^2$ and $f^2 - f^1$ are created within the receiver. They are called 3rd order intermodulation products. They can interfere with the wanted signal like any other signal. A multiply increase of 3rd order and higher order products are created if a large number of signals are in the adjacent bands. The strength of intermodulation products is a function of the source signal strength and the receiver linearity.

Intermodulation can be a problem when many adjacent signals are present. An example is the coexistence of a GSM with a TACS network. Both network technologies use the same uplink and downlink frequency ranges. If a TACS base station with about 20 active channels (each of 10 W TX power) is located at the periphery of a GSM cell, the resulting intermodulation products can cause serious quality degradation in downlink while the uplink is completely unaffected.

Typically, all 3rd party interference effects will have impact – either on downlink or uplink – but not in both directions. Therefore, uplink and downlink measurements are required which is difficult to achieve with RXQUAL without significant effort.

Different radio link topology in uplink and downlink direction

Within traditional networks and network usage with all base stations above roof top level, and all mobile phones on street level (as for car telephone) simulations have shown that uplink and downlink interference levels are quite similar.

Today, many base stations are installed at microcellular sites below roof top. On the other hand, mobile phones are used as hand-held units in any situation from deep indoor at ground level up to elevated positions in high-rise buildings above the level of most of the base stations. This created a variety of scenarios and only one of these is investigated as an example:

Table F:
Case Study
Interference suffered and caused by a single mobile phone above roof level (e.g. hand-held in high rise building or topographical location)

Suffered downlink interference:

Downlink interference suffered by the elevated mobile phone is typically within acceptable order of magnitude due to extremely good serving cell signal. However, signals are not longer ranked by distance r according to $1/r^4$ but to $1/r^3$ ... $1/r^2$ due to virtually free space propagation conditions to the surrounding base stations. This may cause unexpected problems if a co-channel base station is within visibility.

Caused uplink interference:

Significant uplink interference may be caused by the elevated mobile phone. Its uplink (which is similar to free space propagation) competes with the uplink of other mobiles in co-channel cells. Those mobiles are potentially in indoor (-15 dB penetration loss) or deep indoor (-30 dB penetration loss) locations so that the range of power control (-20 dB for a 2W hand-held) may turn out to be insufficient.

Many other scenarios are possible and in most of these cases, uplink and downlink interference mechanisms would be different. Therefore, investigation of differences in uplink and downlink quality in different usage situations (street level, indoor ground level, elevated positions) by use of quality testing tools such as QVoice[3)] is a method to understand the interference mechanisms and to derive conclusions for network tuning.

The near-far effect

A mobile phone at a cell border and another mobile phone in close proximity to the base station are using channels with a small separation, say 2 channel widths. GSM specifies that a C/I for the 2nd adjacent channel of -41 dB is tolerated. On the other hand, the difference in propagation loss between the near and the far mobile phone to the BTS may easily exceed this threshold. Therefore, uplink power control with an additional 20 dB dynamic range for 2 W hand-held units is a mandatory configuration. On the downlink direction, there is no such problem.

In very specific situations, even more than 60 dB propagation loss difference may occur so that the near-far effect could give a small contribution to unbalanced quality in uplink and downlink.

Propagation characteristics

Due to the reciprocity law of electromagnetic wave propagation the physical path loss measured between mobile and base station antennas is identical. This means that the local average of the propagation loss taken over several wave lengths does not vary significantly.

Different locations of Rayleigh fades

Only the location of individual Rayleigh fades is not correlated mainly due to 45 MHz difference between uplink and downlink frequencies. For a stationary mobile without frequency hopping a different quality for the duration of a call may be measurable. However, this effect does not have a general preference to uplink and downlink and therefore, will average out even over small data sets.

Conclusion

In many cases quality testing equipment have reported different speech quality in uplink and downlink direction, and many problems identified that would not have been detected at all while monitoring the downlink only. These would include unbalanced power budget, internal co-channel or adjacent channel interference or 3rd party interference causing blocking or intermodulation.

Argument No. 11: RXQUAL Does Not Consider Acoustical Power Level Variations

A further capability of QVoice[3)] beside of the evaluation of speech quality is the measurement of acoustical power level. Each speech sample of QVoice is initiated with a synchronisation sweep which is transmitted with a constant and well defined power level. The signal may suffer from acoustical power variations when travelling through analogue transmission lines, or when the 64 kbit/s PCM A-law is converted by the speech transcoder into the 13 kbit/s stream. Normally, a conservation of acoustical power is achieved. However, certain GSM operators may decide to modify the power level within the TRAU (Transcoder/Rate Adaptation Units) to cope with the different environmental conditions of mobile phones in comparison with standard phones.

The effect of this tuning measures and the stability of acoustical power level for the duration of a call cannot be monitored with RXQUAL. The measurement is only possible with inband signal strength measurements as provided by QVoice.

Importance of Speech Assessment in QoS

The ultimate criterion for QoS provided by a network operator is the satisfaction of the mobile phone user. Users are sensitive to "network search" periods as well as the time it takes to set up a call. They remember dropped calls especially when the phone's display indicates "full field strength" and the frustration of poor speech quality – where conversations consisted of repeating broken sentences and tolerating an echo. Surveys have shown that listening effort in both directions is the single most important factor for sustained subscriber satisfaction. The key then to high QoS, comes by consistently and accurately evaluating voice quality from the end user's perspective.

Assessing speech quality is a complex subject involving the human perception process and only partly depends on technical parameters which give limited or often misleading information such as discussed in this White Paper on RXQUAL.

There are a number of alternative ways of assessing voice quality, but most evaluation methods were developed to assess quality of speech coders and speech transmission systems under laboratory conditions. QVoice[3)] however, is a unique system – that tests and analyses speech quality as well as network performance – on a real-time basis – at any location within a cellular network service area.

QVoice Introduction

The engineering and consulting group of Ascom Infrasys AG, Solothurn, Switzerland – who are supporting this White Paper Series on cellular network – also developed QVoice – an advanced testing and analysis system designed exclusively to evaluate speech quality on the communication link in cellular networks.

All the shortcomings of RXQUAL presented in the White Paper are resolved with QVoice – including the measurement of each of the Quality of Service parameters listed in the chapter entitled QoS Criteria.

To highlight a few benefits, QVoice can provide the following information unavailable with RXQUAL measurements:

- Report quality improvements for slow moving mobile phones in fading environments after activating frequency hopping – Argument No. 1
- Detect the actual quality degradation due to stolen frames and measure the efficiency of speech frame substitution – Argument No. 2
- Detect echo and other speech degradation due to PSTN analogue effects such as crosstalk or erroneous noise. This is achieved by using specific DSP-based algorithms – Argument No. 3
- Detect ‘Ping-Pong’ sound effects – Argument No. 6
- Measure the quality improvement in interference limited networks after activating frequency hopping; and, assist in necessary BSS parameter revision for handover and power control – Argument No. 7
- Measure actual speech quality following enhanced full rate speech codec introduction without need to re-evaluate meaning of RXQUAL – Argument No. 8
- Compare QoS in networks of different technologies such as analog and digital – Argument No. 9
- Detect uplink quality degradation by 3rd party intermodulation or simple co-channel interference which are invisible on the downlink – Argument No. 10
- Supervise acoustical power level stability for the duration of a call – Argument No. 11

In most situations RXQUAL and Speech quality measurements done by QVoice are in reasonable correlation. However, it is just the special case of discrepancy which need special attention. Two examples are provided to demonstrate that significant inconsistencies between RXQUAL and actual speech quality on the downlink have been observed.

Figure 7a provides a typical case when RXQUAL from 0 to 1 suggests an excellent service. However, the speech quality measurement done by QVoice reports only a "fair" speech quality. Potential causes may be found within the transcoding devices or within the PSTN and further investigation may be suggested after such an observation.

But also the opposite case might be observed. Figure 7 b demonstrates a call where RXQUAL from 3 to 5 suggests marginal quality. However, speech quality measurement done by QVoice reports an 'excellent' speech quality. Those situations are typical for travelling with normal car speed through the Rayleigh fading environment or with frequency hopping as described in Argument No. 1. The network optimization engineer will use this kind of measurements to adjust RXQUAL thresholds which are used in power control and handover algorithms.

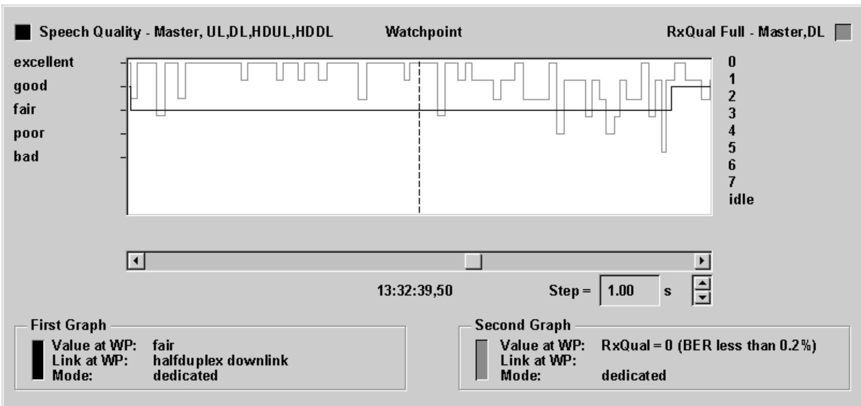

Figure 7a:
RXQUAL is not sensitive to speech quality degradation

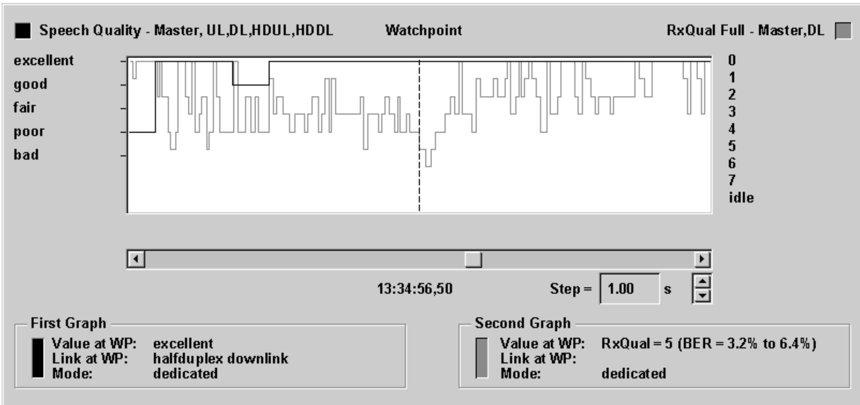

Figure 7b:
RXQUAL does not reflect the high protection of speech transmission by convolutional coding and interleaving

Speech Quality Measurement with PACE

The secret of the system's power is simple: QVoice – in addition to recording conventional air-interface signals – measures speech quality accurately from the user's perspective. It therefore takes into account all degrading effects generated by lack of field strength coverage, co-channel interference, adjacent channel interference, radio multipath delays, blocking due to lack of radio channels, lines or switch capacity, echo, or any other causes. It has many clear advantages over alternative methods.

Since Ascom first introduced QVoice to the market in 1993, based on a then-revolutionary application of patented neural networking technology, it has continued to invest millions of dollars into R&D and into even more advanced techniques for optimising network quality. The result is the Ascom speech evaluation algorithm called PACE.

PACE, which evaluates end-to-end speech transmission quality with real speech samples, and is now being used by Ascom as the basis for both the QVoice production line and its fixed-line network equivalent, QNet. PACE was extensively tested by the International Telecommunications Union (ITU) in September 1998 and found to be the only algorithm that produced excellent results in all experiments, accurately predicted speech quality in the presence of transmission errors and performed equally well for all languages used.

For further detailes on PACE, please refer to the Ascom Technical White Paper "Speech quality and its Objective Evaluation with PACE".

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1) BER exists also in the uplink but is measureable only in the base stations

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