3GPP TR 43.935 V0.3.0 (2005-06)

Technical Report

3rd Generation Partnership Project; Technical Specification Group GSM/EDGE Radio Access Network; Feasibility study of enhanced support for video telephony service over GERAN via the A interface (Release 7)

GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS

Keywords

GSM, GERAN

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Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

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Introduction

The same type of circuit switched (CS) video telephony service as in UTRAN can be provided today in GSM/EDGE Radio Access Network (GERAN) using Enhanced Circuit Switched Data (ECSD). The UTRAN service is using a 64 kbit/s bit transparent radio bearer and for ECSD this is achieved by using two E-TCH/F32 time slots.

When comparing the radio link performance for the circuit switched video telephony service using ECSD and normal speech (enhanced full rate speech) in GERAN there is a substantial difference. This difference can limit the video telephony service to only a part of an existing network.

The interworking between UTRAN and GERAN for the video telephony service is also considered as an important area since video telephony service in GERAN is seen as a complement to the UTRAN service.

The objective of this TR is to provide an overview of the possible enhancements for video telephony in GERAN via the A interface, areas that are considered are coverage, service continuity and quality.

1 Scope

The present document provides an overview of the possible enhancements to the circuit switched vide telephony service in GERAN.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] ITU-T Recommendation G.114, "One way transmission time"

[2] 3GPP TR 45.903, "Feasibility Study on Single Antenna Interference Cancellation (SAIC) for GSM networks"

[3] 3GPP TR 26.912, "Quantitative performance evaluation of H.324 Annex C over 3G"

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions given in 3GPP TR 21.905 and the following apply.

<keyword> <Definition>

3.2 Symbols

For the purposes of the present document, the following symbols apply:

<symbol> <Explanation>

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

CS Circuit Switched
ECSD Enhanced Circuit Switched Data
GERAN GSM/EDGE Radio Access Network
UTRAN Universal Terrestrial Radio Access Network

Other abbreviations used in the present document are listed in 3GPP TR 21.905.

4 Requirements, working assumptions and open issues for enhanced support for video telephony service over GERAN via the A interface

4.1 Requirements

a) Multi-RAT terminals shall be able to perform seamless handover between GERAN and UTRAN for CS video telephony calls.

b) The GERAN support for CS video telephony shall interwork with the current UTRAN video telephony service.

c) The GERAN support for CS video telephony shall support the 3G-324M video telephony standard.

d) The end-to-end delay of the GERAN VT bearer (excluding delay in VT clients) shall not exceed 240 ms in MS to MS video calls.

e) A bit error rate of 10^{-4} shall be defined as a reference performance.

f) The C/I or C/N values required to meet the reference performance shall not exceed the corresponding C/I or C/N values for EFR by more than approximately 6 dB for the least robust specified channel coding scheme.

g) The C/I or C/N values required to meet the reference performance shall be close to the corresponding C/I or C/N values for EFR for the most robust specified channel coding scheme.

h) It shall be possible to perform a fallback to speech from video telephony at bad quality or at handover. The fallback can be initiated either by the network or the mobile station.

i) It shall be possible to run a CS video telephony and a low rate PS service simultaneously.

j) It shall be possible to switch from video telephony to speech on a request from any user involved in the video call.

4.2 Working assumptions

a) The interleaving depth will not exceed 60 ms for all the specified channel coding schemes.

b) A set of channel coding schemes, each one with a specific data rate/TS, will be specified. Each channel coding scheme will provide 64 kbps over a specific number of TSs (one of them could be half TS).

c) Link Adaptation will be provided in order to select the channel coding scheme depending on the radio conditions and the number of TSs depending on the radio resource availability as the best trade-off for providing 64 kbps per video call.

d)

4.3 Open Issues

a) Should lower codec rates than 64 kbps be supported by GERAN VT bearers?

b) Should a service change from speech to video telephony be possible?

c) What is the necessary range of possible number of TSs assigned to a video call (not necessarily all the assigned TSs are actually (on a per TDMA frame basis) used during a video call)?

5 Quality aspects

5.1 ECSD/3G-324M media delay

In ITU-T recommendation G.114 [1] the effects of one way transmission delay on a telephone connection is described. Although G.114 is dealing with telephone (i.e. speech only) connections it is reasonable to use it for the audio part of video telephony as well.

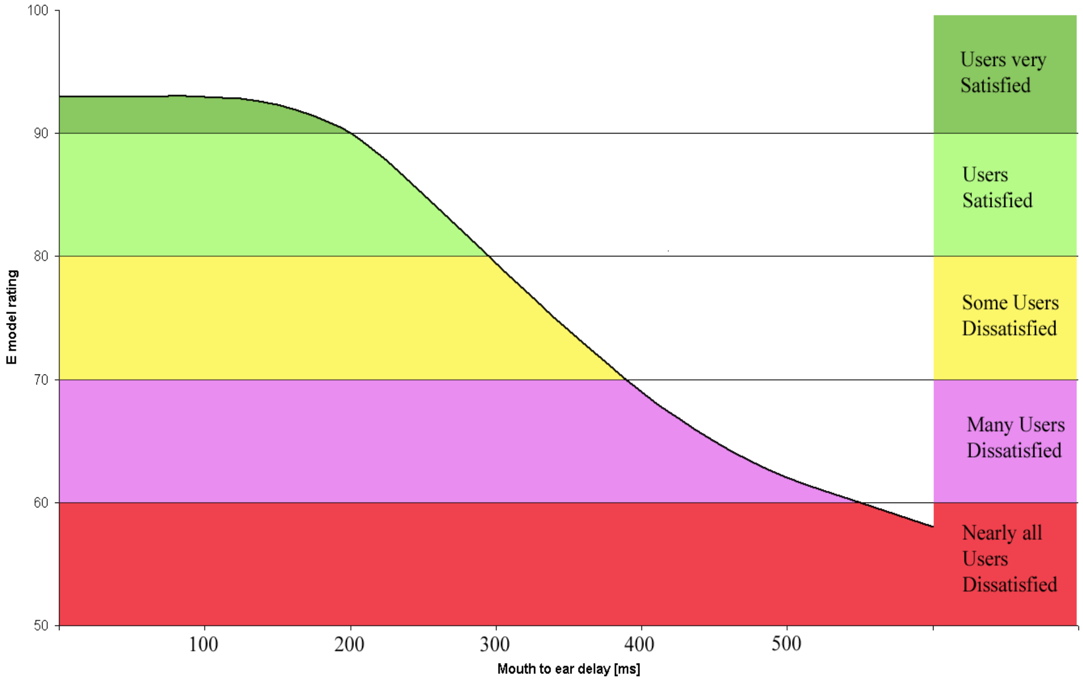


Figure 1. Quality versus delay.

Figure 1 shows how the end user perceives the connection quality as a function of one-way mouth to ear delay according to G.114. The main reason why a long delay is not acceptable is that it becomes hard to switch direction in a conversation and the two parties will talk at same time.

All speech services in GSM (i.e. full-rate, enhanced full-rate, AMR full-rate etc) have a one way delay of approximately 200 ms in MS to MS calls which, according to figure 1, will not degrade the quality.

It is important that new suggested improvements for enhanced support for CS video telephony does not make the one-way delay exceed 400 ms.

6 Coverage aspects

6.1 Coverage and link performance for CS video telephony

There are differences between the radio link performance for CS video telephony service using the 64 kb/s transparent bearer in ECSD and full rate speech (EFR). The main reason for this difference is that the ECSD timeslots are transporting almost 3 times the number of bits per timeslot compared to the full rate speech service. The higher bitrate for the ECSD timeslots implies a higher sensitivity to radio disturbances like noise and interference.

During the work on video telephony for WCDMA 0.2-0.3% block error rate per link has been considered to give good video quality, see [3]. The C/I or C/N required to achieve this with E-TCH/F32 (interleaving depth corresponding to 60 ms) is approximately 6 dB higher than the C/I or C/N necessary to give a corresponding quality level for full rate speech (EFR).

However, in a coverage study performed by Ericsson for a real radio network it is shown that the coverage still can be rather good in most areas except for rural. The details for this study are presented in section 6.1.1.

6.1.1 Coverage study

This coverage study has been performed by using an Ericsson developed coverage prediction tool called TEMS Cell Planer Universal 5.1. It is the downlink coverage that is considered with this tool. Input to this tool has been parameters from a real network and assumptions on the signal strength requirement and other relevant link budget parameters. The figures on the parameters can be found in table 1 and table 2.

There are three different figures on the assumption for signal strength requirement on video telephony, and this is because there are some extra factors that can reduce video telephony coverage compared to full rate speech (EFR).

These factors are:

- **Link performance.** The requirement on C/I or C/N is approximately 6 dB higher for video telephony compared to full rate speech (EFR).
- **Lower average out put power.** For a given maximum amplitude the average transmission power in 8-PSK will be lower than in GMSK due to the peak to average variations. This difference between GMSK and 8-PSK is ~3dB.
- **Transmitter linearity.** 8-PSK modulation has a varying envelope which imposes stricter linearity requirements on the transmitter than GMSK. Depending on transmitter design, it may be necessary to “back-off” and transmit 8-PSK with lower power than GMSK.
- **Mobile power reduction.** It is permitted in 3GPP 45.005 for terminals to reduce the output power when transmitting on more than one timeslot. The main reason is to control heat dissipation. For two time slot transmission the permitted power reduction is 3 dB. This reduction is not mandatory, and it might be possible to keep the same output power per timeslot when transmitting on two timeslots.

There is also another factor in the link budget that is different between full rate speech (EFR) and CS video telephony and that is:

- **Body loss.** In a link budget a so-called “body loss” factor is included. This factor models the attenuation caused by the body of the user. For video telephony in WCDMA the body loss factor is set to zero since it is expected that a video user will hold the terminal in the hand instead of close to the head which will reduce the body loss.

Table 1 summarizes three different combinations of these factors. The background for the three cases is:

- **Best case.** In this case there are no differences in output power for GMSK and 8-PSK. This can be realistic in a scenario with short site-site where it is not needed to use full output power.
- **Linear transmitter.** In this case there are lower average output power for 8-PSK compared to GMSK due to the peak to average variations for 8-PSK. Linear transmitter sending GMSK at full power but no penalty for sending on two timeslots. Realistic in downlink and may be realistic in some in terminals.
- **MS transmitter.** Same as “Downlink” but with an additional 3 dB lower power for the linearity aspect to set a case more realistic for the uplink. No additional power attenuation has to be added for “MS power classes” since the transmission power is already 6 dB less than for GMSK due to the other factors.

Table 1 Speech – Video link budget difference

Factor	Best case	Linear transmitter	MS transmitter
Link performance	- 6 dB	- 6 dB	- 6 dB

Peak/Average	0 dB	- 3 dB	- 3 dB
Linearity	0 dB	0 dB	-3 dB
MS power classes	0 dB	0 dB	0 dB
Body loss	+ 3 dB	+ 3 dB	+3 dB
Total:	**- 3 dB**	**- 6 dB**	**- 9 dB**

These three cases (or a subset) are used in the coverage analysis in the following sections.

Other relevant parameters that have been used in the simulations are described in table 2.

Table 2 Parameters used in the simulations

Antenna	65 degrees, 16.5 dBi
Mobile sensitive	-104 dBm
Raleigh fading margin	3 dB
Interference margin	2 dB
Traffic	Uniform
Number of cells, dense urban	105
Number of cells, suburban	117
Dense urban map	- 5m resolution in path loss predictions - Main clutter types: building and open - GSM-900 urban prediction model
Suburban map	- 50 m resolution in path loss predictions - Main clutter types: suburban, trees forest, lake etc. -9999 prediction model -LNF outdoor margin 95% coverage probability: 3 dB -LNF indoor margin 95% coverage probability: 6.8 dB - Building penetration loss: 12 dB

6.1.1.1 Urban environment

Parameters from a real radio network in a dense urban environment are used in this section. The coverage prediction is made for 900 MHz.

Only macro sites were used in the predictions, no micro cells and no dedicated indoor systems. The site to site distances are in the range of 100-800m. The map used contained detailed building data with a resolution of 5 m but no windows. The main coverage holes seen in the simulations were in large buildings that may be covered by dedicated in-building systems or from outdoor through the windows.

Table 3 Coverage for GSM900 in a dense urban network

Service	Speech	Video (Best case)	Video (Linear transmitter)	Video (MS transmitter)
Signal Strength Requirement	-96 dBm	-93dBm	-90 dBm	-87 dBm
Indoor Coverage	97.8%	96.8%	95.4%	93.4%
Outdoor Coverage	100%	100%	100%	100%

In Table 3 the results of the predictions is shown. They show a loss in indoor coverage for video compared to speech but the loss is quite small. It can be discussed what level of coverage should be considered acceptable. It is reasonable to accept some loss. If not it can be fixed with a limited number of new sites.

6.1.1.2 Suburban environment

Some downlink predictions have also been made for a suburban area with site to site distances of 500m to more than 3 km. For a coverage probability of 95%, i.e. a 95% probability to have a signal strength exceeding the required value within the coverage, the indoor coverage for video can in most cases be considered acceptable. The video coverage holes can be taken care of with a limited number of new sites. The 95 % level is a planning criteria often used.

A suburban network is generally planned with less coverage margins compared to an urban network as the requirements on capacity and indoor coverage is less stringent. This may result in that video coverage is more of an issue than in an urban network.

Table 4 shows the result of the suburban predictions. Note that the difference in signal strength requirements between the urban and suburban prediction is due to the lack of building data in the suburban predictions. A building penetration loss and a Log Normal Fading (LNF) margin have to be added in the suburban case.

Table 4 Coverage for GSM900 in a suburban network

Service	Speech	Video (Best case)	Video (Linear transmitter)	Video (MS transmitter)
Indoor Signal Strength Requirement	-77.2 dBm	-74.2dBm	-71.2 dBm	-68.2 dBm
Indoor Coverage	97.6%	93.2%	85.6%	74.6%
Outdoor Signal Strength Requirement	-93.0 dBm	-90 dBm	-87 dBm	-84 dBm
Outdoor Coverage	100%		99.95%	

6.1.1.3 Rural environment

For the rural area (site to site distances of 10 to 30 km) the speech coverage was already limited which in turn means that the video coverage was far from acceptable.

It is however obvious that we will see the largest coverage problems at the country side where the link budget for speech is stretched and where there is less “over coverage” than in the central part of a city. Interference and limited spectrum resources, on the other hand, might not be a problem at all as the traffic load is rather low in rural areas.

6.1.2 Considerations for ECSD in a real network

Due to the relatively high interleaving depth (corresponding to 60 ms) and relatively low code rate (0.468), E-TCH/F32 performance is better with frequency hopping than without. For example, the difference between ideal frequency hopping and no frequency hopping in a TU3 environment at 900MHz exceeds 10 dB in noise, cochannel interference, and adjacent channel interference. Moreover, the absolute C/I values needed to achieve 0.25% block error rate in TU3noFH exceed 25 dB. Such high C/I ratios would be difficult to achieve over wide areas and very expensive in terms of radio resources. The conclusion is that a video telephony service over ECSD must be placed in a frequency hopping channel group.

Even with frequency hopping, E-TCH/F32 is considerably less robust than EFR and AMR speech at their appropriate operating points. The loss in sensitivity and interference tolerance compared to EFR is about 6 dB. This naturally translates into a loss of coverage in noise-limited environments. In interference-limited environments, a separate “high quality” frequency hopping channel group with a sparse reuse and/or low frequency load is needed if the video telephony and traditional telephony services are to be available over similar areas.

6.1.3 Improvements to the link performance for CS video telephony

As seen in previous chapters there is a difference in radio link performance between CS video telephony and full rate speech (EFR), especially in rural areas. To provide the same coverage for CS video telephony as for full rate speech (EFR) in rural areas, there will be a trade-off between coverage and quality or capacity.

The main items when improved link performance for CS video telephony is discussed are:

- New channel coding, this will probably not improve the link performance enough for the rural coverage.
- Improved receiver performance, there might be receiver improvements that can be made for the CS video telephony service.
- Use more TSs for the CS video telephony service over the air interface. This will have negative impact on the capacity and the possible reduction of the output power from the MS can limit the benefit with this. This might also require MSs that can transmit and receive at the same time.
- Decrease the quality of the CS video telephony service in areas with bad coverage to maintain the connection. Examples on this are to use 32 kb/s transparent bearer (this is still an open issue) when the quality for the 64 kb/s transparent bearer is too bad or to switch to normal speech in these areas. This will have negative impact on the quality and may give interoperability problems with WCDMA.

6.2 Link and system level performance of improved channel coding schemes

The link and system level performance of several improved channel coding schemes for CS video telephony are evaluated in this section.

6.2.1 Evaluated schemes

The considered options for improving the coverage of the existing VT bearer (2 x ECSD/32.0k) are the following:

- Increased resource bandwidth (more timeslots)
- Decreased codec rate (32 kbit/s)
- Link adaptation

6.2.1.1 Increased resource bandwidth

The resource bandwidth of VT can be increased by using more time slots. If the payload (codec rate) remains the same, the extra slots can be used to decrease the channel coding rate and hence improve the coverage.

In practice, the number of available time slots is limited by the multislot class and uplink power restrictions. Furthermore, the maximum uplink power of 8PSK is lower than the maximum power of GMSK, since the PA cannot operate in its saturation region.

The evaluated channel allocation schemes are briefly described below. It is assumed that the modulation is 8PSK and the 3G-324M codec produces a bitstream of 64 kbit/s.

- *2DL+2UL*. The 2/2 scheme provides two E-TCH/F32.0 channels for the transmission of VT traffic. The E-TCH/F32.0 was modified in GERAN#22 in order to reduce the interleaving delay of the VT bearer and to make the channel coding FLO-compatible. In this study, the 2/2 scheme will be used as a reference (worst) case.
- *3DL+3UL*. The 3/3 scheme provides a symmetric VT bearer with increased robustness for both uplink and downlink. Unfortunately, with three slots in uplink, it is very likely that the MS cannot transmit at full power. If it is assumed that no power reduction is needed with the transmission of two uplink 8-PSK slots, the transmission of three slots would imply a reduction of 1.8 dB (at maximum).
- *3DL+2UL*. The 3/2 scheme can be seen as a compromise between the symmetric schemes. If the link balance was limited by the downlink, the performance of the 3/2 scheme is potentially close to the 3/3 scheme.
- *2DL+3UL*. The 2/3 scheme would be useful in networks, where the link balance is limited by the uplink. Typically, this condition is valid in rural area deployments. The main downside of the 2/3 scheme is that the gain obtained from the more robust uplink coding is partly lost for the reduced mobile power (up to 1.8 dB).
- *2.5DL+2.5UL*. If a symmetric allocation was needed, but the 3/3 scheme is not feasible, the 2.5/2.5 scheme could be an option worth considering. The 2.5/2.5 allocation means that the time slot configuration is alternated between the 3/2 and 2/3 schemes. The main benefit of this scheme is that the uplink power needs to be decreased only by ~1 dB in contrast to the ~1.8 dB reduction for the 3/3 scheme. The major downside of the 2.5/2.5 scheme is that it would likely require more implementation work than the other schemes.

One benefit of the 3/2, 2/3, and 2.5/2.5 schemes over the 3/3 scheme is that a mobile satisfying *sum=5* (MSC 8-12) can be used. Today, it is not possible to use a MSC greater than 18 for ECSD/HSCSD.

6.2.1.2 Decreased video codec rate

The sensitivity of the VT bearer can be also increased by decreasing the video coding rate. In this study, the performance of a 32.0 kbit/s bearer will be evaluated. Such bearer would be capable of providing, for example, 4.75 kbit/s for the AMR speech, ~24 kbit/s for the MPEG-4 video, and ~4 kbit/s for the MUX overhead. It is here assumed that the current generation of low-rate video codecs is capable of providing sufficient quality on a 32 kbit/s bearer.

One option would be the use of GMSK modulation for the most robust channel coding mode (at cell borders). The channel coding of this scheme would be less robust than the channel code of the corresponding 8-PSK configuration, but this impairment would be compensated by the higher maximum output power and the higher sensitivity of the GMSK modulation. Furthermore, it would be possible to apply SAIC at the mobile side. It should be noted that the interleaving engine of FLO does not currently support the 60 ms interleaving on GMSK channels.

6.2.1.3 Link adaptation

In order to have the maximum benefit from the more robust channel coding schemes, the VT should support link adaptation. The considered LA modes are shown in table 5 below:

Table 5 Link adaptation modes

LA mode	**Codec mode**	**Channel allocation**	**Coderate**
1	64 kbit/s	2DL / 2UL	0.46DL / 0.46UL
2	64 kbit/s	2.5DL / 2.5UL	0.37DL / 0.37UL
3	64 kbit/s	3DL / 2UL	0.31DL / 0.46UL
4	64 kbit/s	2DL / 3UL	0.46DL / 0.31UL
5	64 kbit/s	3DL / 3UL	0.31DL / 0.31UL
6	32 kbit/s	2DL / 2UL	0.23DL / 0.23UL

It could be, for example, possible to use the LA modes 1 and 3 in the urban areas, where the network is typically DL limited. On the other hand, modes 1 and 4 could be used in rural areas, where the link is often limited by the uplink. The 32 kbit/s mode could be used e.g. to enhance the coverage at the cell borders and to fill the gaps in indoor coverage.

6.2.2 Link simulations

6.2.2.1 Configuration

The link level performance of the studied channel coding schemes was evaluated with a proprietary GSM/EDGE link lever simulator. The following common parameters were used for all simulations:

Table 6 Common parameters for the link level simulator

Parameter	**Value**
Number of simulation steps (per CIR value)	200000
Channel model	TU3iFH
Frequency band	900 MHz
Interference	Co-channel
Receiver impairments	Phase noise Mixer phase error I/Q amplitude imbalance DC offset

The channel coding of the evaluated LA modes was carried out with the Flexible Layer One. The transport block sizes were determined by evenly splitting the codec payload into appropriate number of FR time slots. The only exception was the LA mode 2 (2.5/2.5), where the VT traffic was assumed to be carried with two FR-channels and one HR-channel. The transport block size was 512 bits for the full rate channel, and 256 bits for the half rate channel. The BER for the LA mode 2 was then obtained from

$$(1/2.5)\cdot\left(2\cdot BER_{FR}+0.5\cdot BER_{HR}\right),$$

where BER_{FR} and BER_{HR} represent the bit error ratios of FR and HR configurations, respectively.

All other FLO parameters were the same among the LA modes. The parameters are summarized in table 7.

Table 7 FLO configuration parameters

Parameter	LA mode								
	1	2		3		4		5	6
		FR	HR	DL	UL	DL	UL		
Transport block size (bits)	640	512	256	427	640	640	427	427	320
CRC [bits]	0	0	0	0	0	0	0	0	0
TFCI [bits]	0	0	0	0	0	0	0	0	0
In-band signalling [bits]	0	0	0	0	0	0	0	0	0
Rate matching attribute	n/a	n/a	n/a	n/a	n/a	n/a	n/a	n/a	n/a
Modulation	8PSK	8PSK	8PSK	8PSK	8PSK	8PSK	8PSK	8PSK	8PSK
Channel mode	FR	FR	HR	FR	FR	FR	FR	FR	FR
Interleaving depth [ms]	60	60	60	60	60	60	60	60	60

The channel coding for the enhanced full rate speech (TCH/EFS) was taken as a reference case. The EFR simulations were configured with the same general parameters as the FLO simulations (see table 6).

6.2.2.2 Results

According to the agreed service requirements, the average bit-error-rate of a VT service shall be less than 10^{-4}. Furthermore, the C/I values required to meet the reference performance shall not exceed the corresponding C/I values for EFR by more than approximately 6 dB for the least robust specified channel coding scheme. It is here assumed that that the EFR performance should be evaluated at FER = 1%.

The simulation results are summarized in table 8, which shows the C/I values of the evaluated LA modes at BER=10^{-4}. The C/I values are compared with the EFR performance (C/I at FER=1%), the reference value being 8.53 dB. In addition, the gains versus the existing VT bearer (LA mode 1) are shown. The BER chart illustrating the performance of different LA modes can be found in Annex A.

Table 8 Link level results

LA mode	CIR@BER = 1e-4 [dB]	Loss versus EFR [dB]	Gain versus mode 1 [dB]
1 (64k, 2/2)	14.27	5.74	-
2 (64k, 2.5/2.5)	13.23	4.70	1.04
3 (64k, 3/2)	10.72 / 14.27	2.19 / 5.74	3.55 / 0.00
4 (64k, 2/3)	14.27 / 10.72	5.74 / 2.19	0.00 / 3.55
5 (64k, 3/3)	10.72	2.19	3.55
6 (32k, 2/2)	9.11	0.58	5.16

As can be seen, the gain compared to the current VT bearer (LA mode 1) ranges between 1.04 and 5.16 dB. While the EFR performance is not reached with any of the schemes, the gap is only 0.58 dB when the 32k codec is applied.

It should be noted that the 2.5/2.5 scheme is slightly sub optimal, since the performance of the HR channel is worse than the performance of the FR channel. Although this problem could be fixed by allocating fewer bits to the HR channel, no significant gain to the present figures is expected.

6.2.3 Network simulations

The network performance was studied with a dynamic high resolution GSM/EDGE system simulation tool. The simulator uses non-averaged burst level mapping between system and link simulators.

6.2.3.1 Configuration

The main simulation parameters are summarized in table 9. They are based on configuration 1 of SAIC feasibility study [2]. The simulated network had hexagon-shaped cells, grouped by three into one site. The transceiver antenna was located in the site center facing outwards. In TCH layer random frequency hopping with 1/1 reuse plan was used. Due to MAIO management, there was no co-channel and no adjacent channel interference between cells in one site. BCCH layer was not included in the simulations.

Table 9 System level network simulation parameters

Parameter	Value and unit	Comment
frequency	900 MHz	
bandwidth	7.8 MHz	for TCH transceivers
hopping	random RF	neither co-, nor adj.ch. interf.
cell radius	500 m (TU), 10 km (RA)	
BTS antenna height	15 m (TU), 30 m (RA)	
number of cells	75	25 sectorized sites
cells per site	3	
slow fading standard deviation	6 dB	
lognormal corr. distance	110 m	
adj. channel interf. attenuation	18 dB	
handover	quality and level based	HO margin 3 dB
power control	quality and level based	
max. BTS output power	31.6 W (45 dBm)	
fast fading	typical urban (TU) channel, rural area (RA) channel	as defined in GSM spec. 05.05
MS speed	3 km/h	
mean call length	90 s	
min. call length	5 s	

Load of the network was chosen so, that the average load of the transceivers in the base stations was around 50%. For the asymmetric schemes in cases 3 and 4 (3/2 and 2/3) the load was about 43% because two slots per TRX in one direction cannot be allocated.

Simulations were run for two kinds of networks:

- urban (TU), 500 m cell radius,
- rural area (RA), 10 km cell radius.

6.2.3.2 Results

The results shown in table 10 contain service probability, when different LA schemes are used. The service probability is defined as the proportion of calls with mean decoded BER less than 10^{-4}. See Annex B for more detailed results.

Table 10 Summary of system-level simulation results

LA mode	Service probability in TU		Service probability in RA	
	Downlink	Uplink	Downlink	Uplink
1 (64k, 2/2)	70.0%	77.3%	70.4%	59.2%
3 (64k, 3/2)	94.1%	88.4%	94.3%	64.6%
4 (64k, 2/3)	86.8%	95.3%	83.1%	76.6%
5 (64k, 3/3)	93.6%	95.1%	94.8%	76.5%
6 (32k, 2/2)	96.2%	97.7%	97.7%	89.7%

The 2/3 scheme performs in DL better than 2/2 because there are two timeslots in DL direction, which cannot be allocated and therefore there is less interference than in 2/2 case. On the other hand in UL, 2/3 performs better than 2/2 because of more robust coding. The same is true for 3/2 but in reversed directions.

In case of 3/3 scheme, there was simulated background speech service to fill the remaining two slots in a transceiver, when it had allocated two ECSD calls (6 timeslots). These speech calls were simulated to add interference in the network, so that there were similar radio conditions as in 2/2 case.

6.2.4 Summary for link and system level simulations

This section has evaluated the performance of five performance enhancement schemes for GERAN video telephony. According to the simulation results, the most potential schemes are:

- Symmetric 2/2 configuration. This scheme is already available and should be used when the signal to noise ratio is at least 6 dB above the planned minimum.
- Asymmetric 2/3 and 3/2 configurations. These schemes would enhance the performance in network scenarios, where the link balance would be in favor of uplink/downlink, respectively.
- Lower video coding rate (32 kbit/s). This scheme would provide a link layer performance that is only 0.5 dB worse than the performance of EFR.

The common denominator for the above-mentioned schemes is that terminals fulfilling the *sum=5* multislot requirement could be used. In order to gain the maximum benefit, link adaptation should be applied between the schemes.

7 Service Continuity aspects

In areas covered by GERAN, service continuity will be guaranteed by adopting link adaptation, i.e. using channel coding schemes with lower code rates and assigning a higher number of TSs to the video call, whenever required by the changed radio conditions.

Based on the coverage study reported in Sect. 6.1.1, the worst link budget difference between video and speech applies for the MS transmitter case, where such difference is estimated to be 9 dB. In such a case, the use of the most robust specified channel coding scheme would imply that the C/I or C/N values are close to the ones required for EFR [according to requirement g) in Sect. 4.1]. In turn, that implies the link performance factor is roughly equal to 0 dB, leading to a residual link budget difference between video and speech of 3 dB. Such a gap may be further reduced by adopting receiver improvements (according to the 2nd bullet in Sect. 6.1.3).

In case of inter-RAT handover (e.g. passing from an area covered by UTRAN to an area covered by GERAN only or performing such handover for load reasons), service continuity will be guaranteed via seamless handover between GERAN and UTRAN and inter-working between GERAN and UTRAN video telephony services [according to requirements a) and b) in Sect. 4.1].

Another way of service continuity is the fallback from video telephony to speech in the following cases:

- bad quality
- handover
- user's request

[according to requirements h) and j) in Sect. 4.1].

8 Technical Solutions

The technical solutions presented in this section are only proposals and a recommendation for which technical solutions that should be included in the 3GPP standard can be found in section 9.

8.1 New channel coding schemes

Besides E-TCH/F32, a set of new channel coding schemes will be specified, each one providing 64 kbps over a specific number of TSs, among which one could be half TS [according to working assumption b) in Sect. 4.2].

Any new specified channel coding scheme will provide a lower code rate and require a higher number of TSs to be assigned to the video call compared to E-TCH/F32, to guarantee service continuity, as highlighted in Sect. 7.

Among the possible new channel coding schemes, 2 proposals are reported hereafter:

- Alternating 3/2 + 2/3 multislot configuration
- Shifted 3 + 3 multislot configuration

8.1.1 Alternating 3/2 + 2/3 multislot configuration

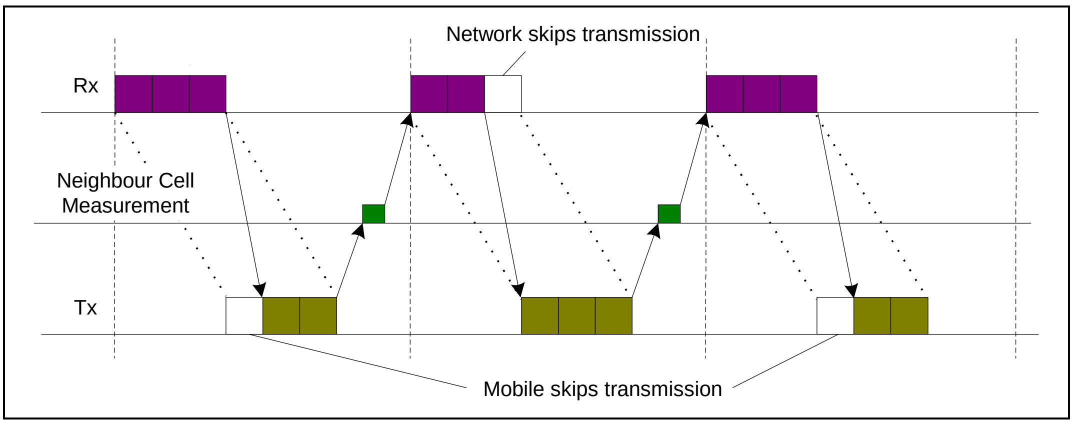


Figure 2: Alternating 3/2 + 2/3 multislot configuration

Figure 2 represents the general principle of the alternating between 3+2 and 2+3 timeslot configuration proposal. The rate at which the mobile station switches between the 2 configurations has to take into account the interleaving depth and is FFS (modifications may also be required to the interleaving on this new half timeslot allocation to avoid introducing any additional delay to the connection).

This multislot configuration is compliant with the multislot capability of a mobile station requesting $T_{tb} = 1$ and $T_{ra} = 2$, but this means that half a timeslot is wasted on an average for the allocation. Whether this half a timeslot could be used for a different service is FFS.

A new channel coding scheme of 25.6 kbps/TS (64 kbps/2.5 TS = 25.6 kbps/TS) is needed to maximize the benefits of the additional half timeslot.

8.1.2 Shifted 3 + 3 multislot configuration

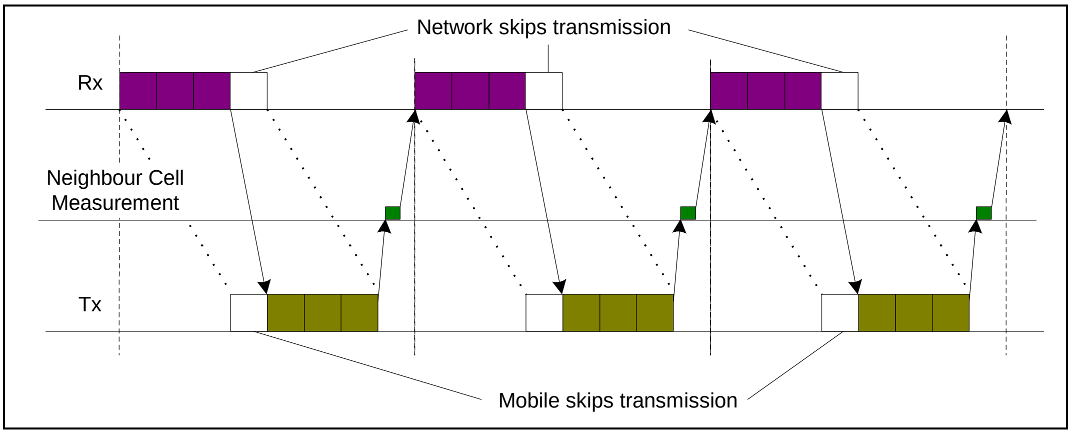


Figure 3: Shifted 3 + 3 multislot configuration

Figure 3 represents the general principle of the shifted 3 + 3 configuration where the mobile station is allocated 4 timeslots on both the uplink and the downlink, but it only transmits and receives on 3 of the 4 timeslots.

This multislot configuration is compliant with the multislot capability of a mobile station requesting $T_{tb} = 1$ and $T_{ra} = 1$, but this means that a whole timeslot is wasted for the allocation. Whether this timeslot could be used for a different service is FFS.

A new channel coding scheme of 21.4 kbps/TS (64 kbps/3 TS = 21.4 kbps/TS) is needed to maximize the benefits of the additional timeslot.

8.2 Link Adaptation

According to working assumption c) in Sect. 4.2, Link Adaptation will be provided in order to select the most suitable channel coding scheme depending on:

- radio conditions,
- radio resource availability.

The selection has to be performed as the best trade-off for providing 64 kbps per video call.

Table 11 provides the general principle of a qualitative selection of the channel coding scheme among E-TCH/F32 and the 2 proposals reported in Sect. 8.1.

Table 11: Qualitative selection of the channel coding scheme

Selected channel coding scheme	Radio conditions	Radio resource availability
E-TCH/F32	good	limited
25.6 kbps/TS	worsen	increases
21.4 kbps/TS	bad	guaranteed

9 Summary and Conclusions

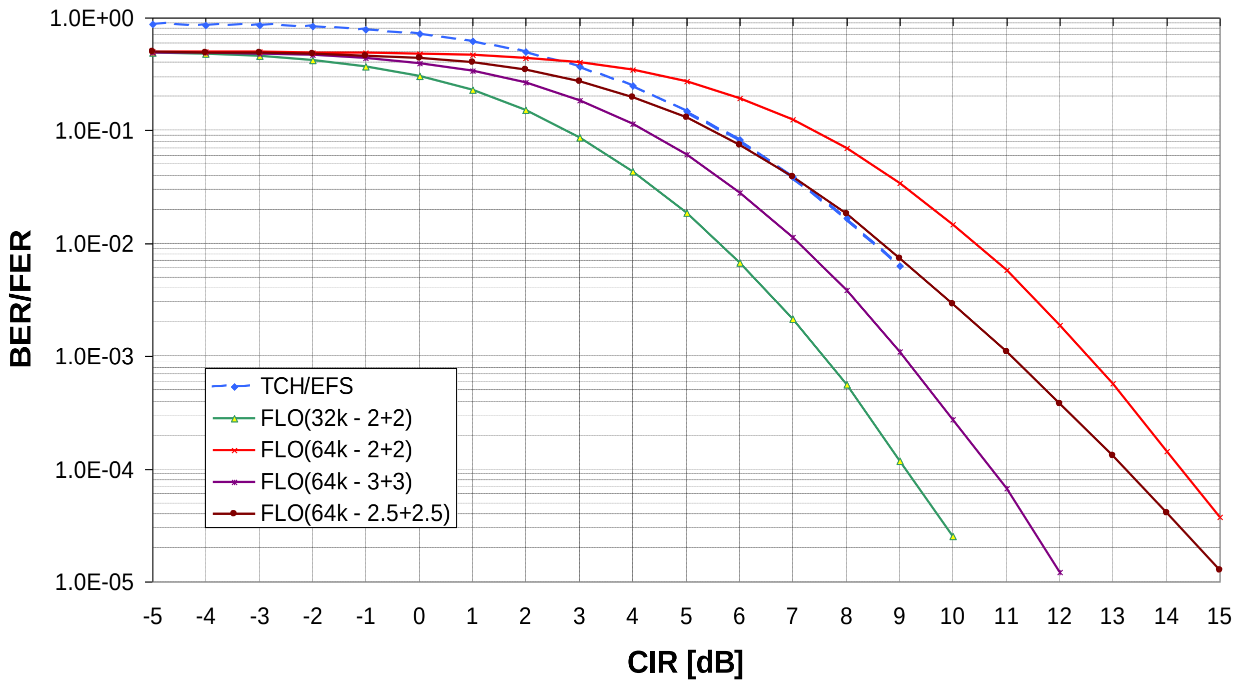
9.1 Summary of open issues

< NOTE: This clause will be a summary of the open issues in this report. >

9.2 Conclusion

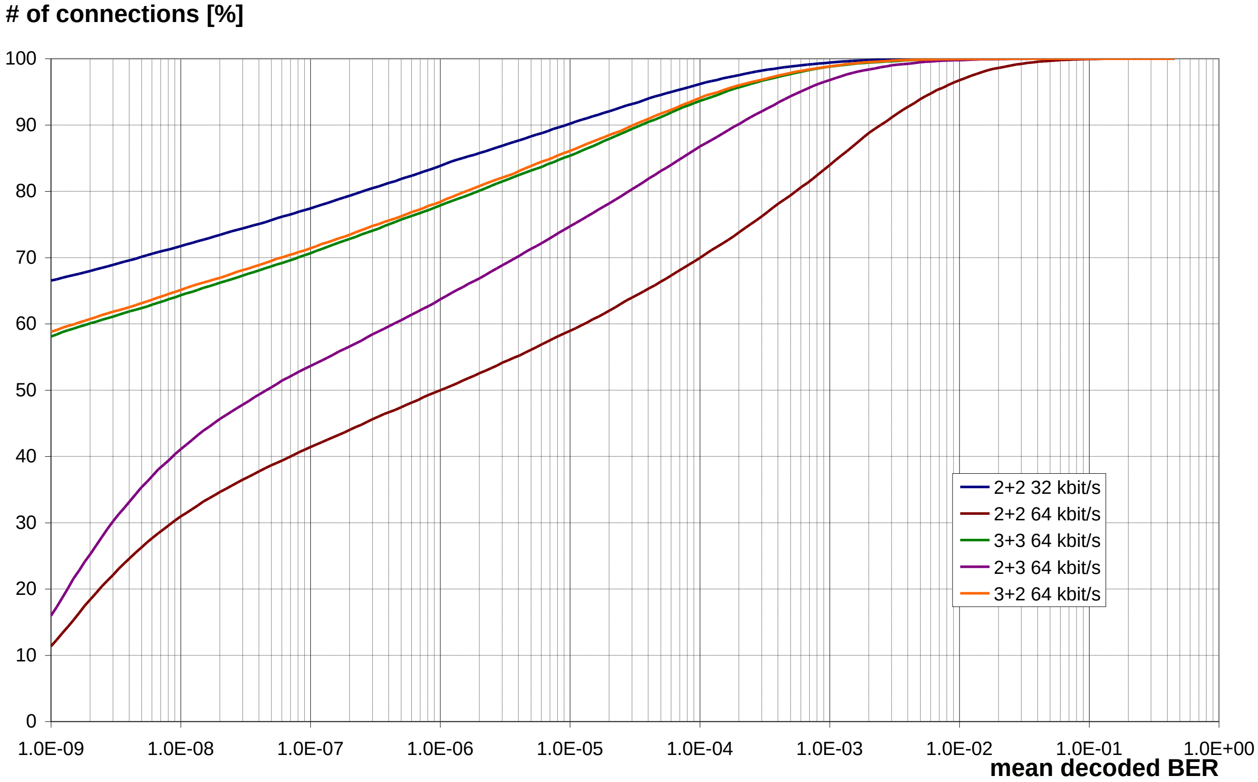
< NOTE: Conclusions drawn from previous chapters will be included in this clause. >

Annex A: Link level results

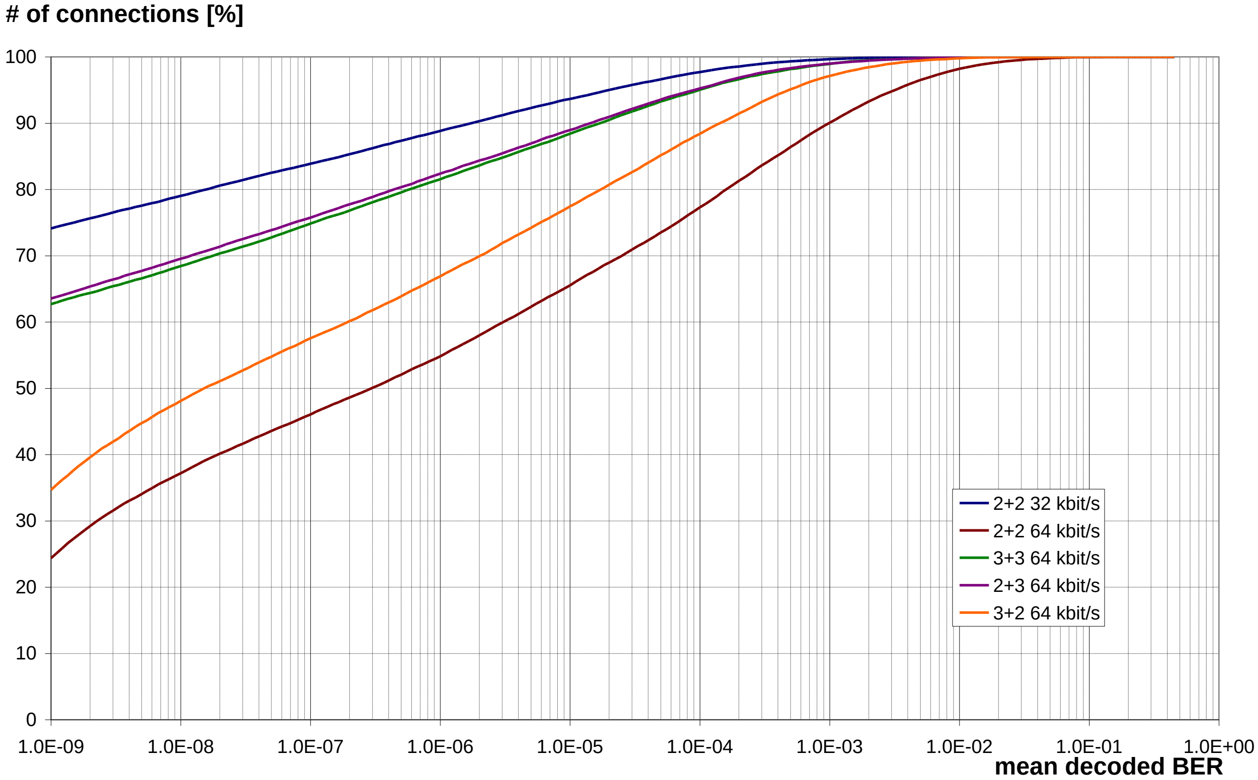


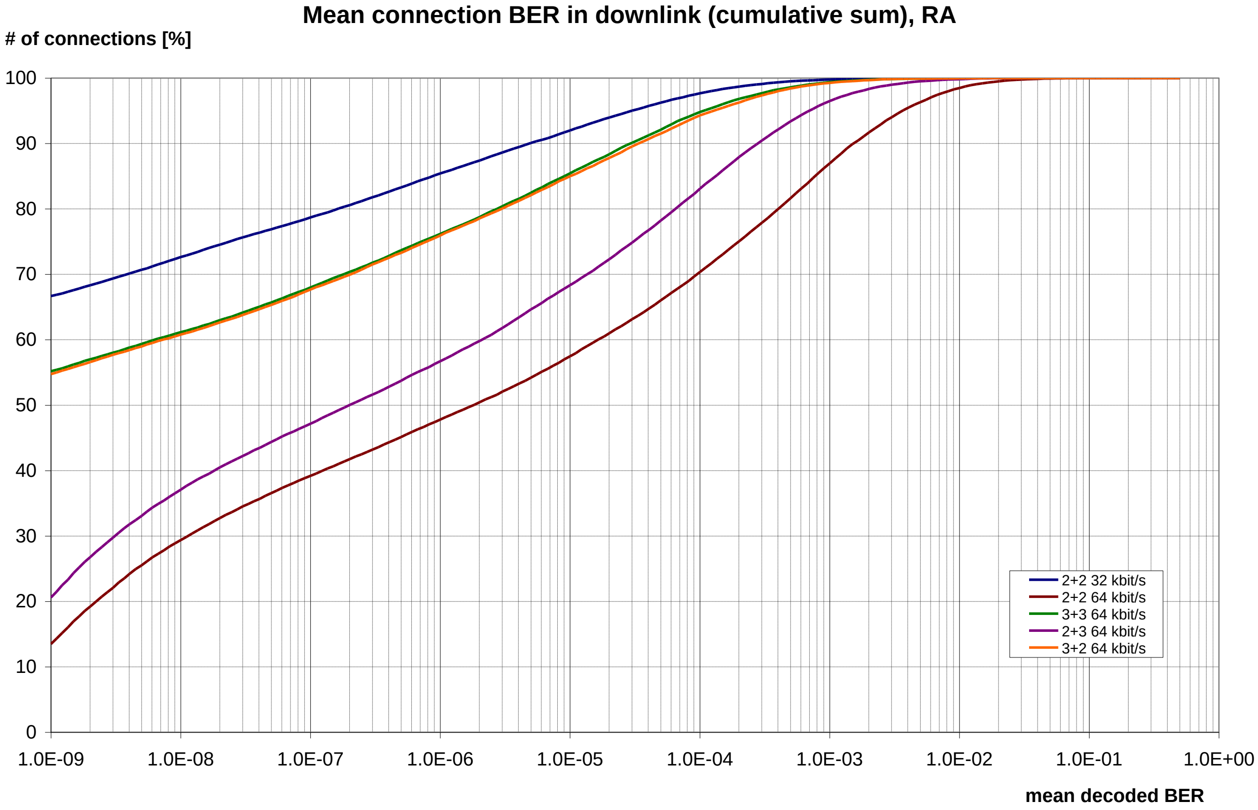
Annex B: System level results

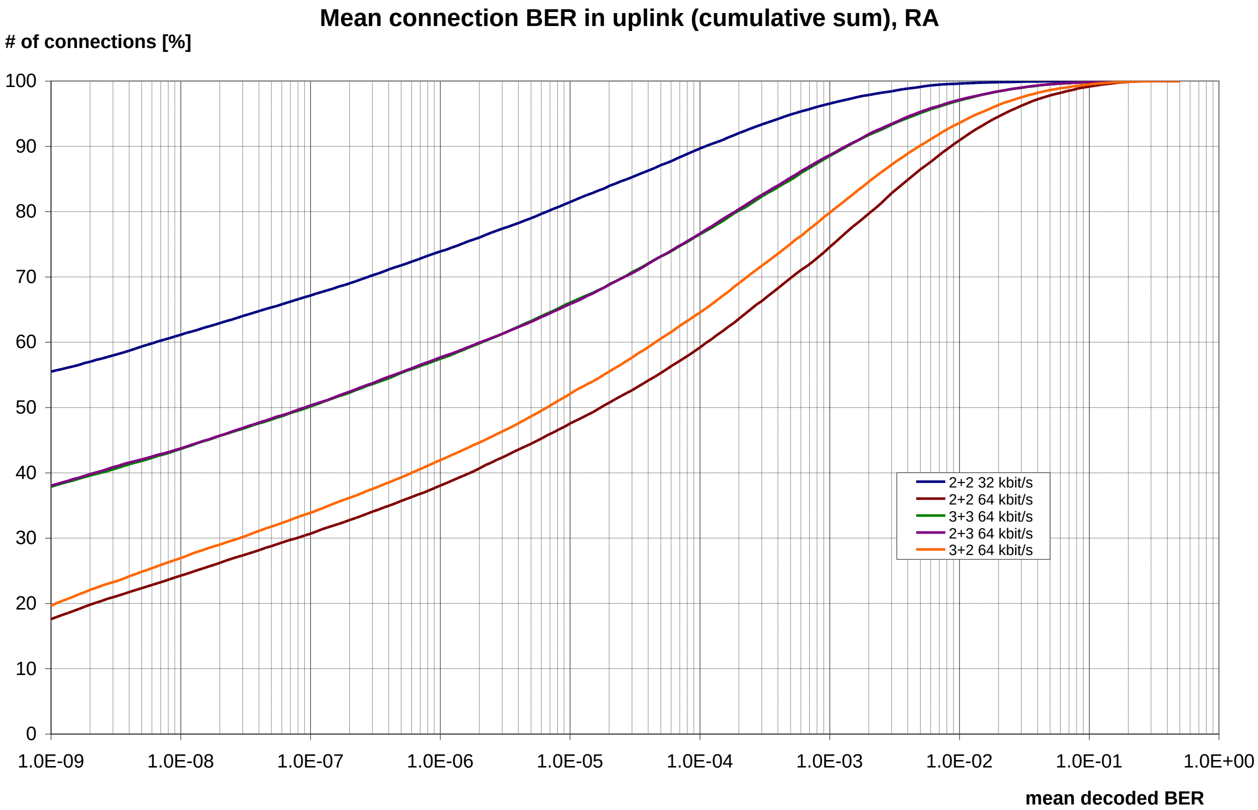
Mean connection BER in downlink (cumulative sum), TU



Mean connection BER in uplink (cumulative sum), TU




Mean connection BER in downlink (cumulative sum), RA
# of connections [%]
100
90
80
70
60
50
40
30
20
10
0
1.0E-09
1.0E-08
1.0E-07
1.0E-06
1.0E-05
1.0E-04
1.0E-03
1.0E-02
1.0E-01
1.0E+00
mean decoded BER
2+2 32 kbit/s
2+2 64 kbit/s
3+3 64 kbit/s
2+3 64 kbit/s
3+2 64 kbit/s


Mean connection BER in uplink (cumulative sum), RA
# of connections [%]
100
90
80
70
60
50
40
30
20
10
0
1.0E-09
1.0E-08
1.0E-07
1.0E-06
1.0E-05
1.0E-04
1.0E-03
1.0E-02
1.0E-01
1.0E+00
mean decoded BER
2+2 32 kbit/s
2+2 64 kbit/s
3+3 64 kbit/s
2+3 64 kbit/s
3+2 64 kbit/s

Annex C: Change history

Change history							
Date	**TSG GERAN#**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2004-11	22				Text incorporated from GP-042532	0.0.1	0.1.0
2005-01	23				Requirements, working assumptions and open issues are added from GP-050542	0.1.0	0.2.0
2005-06	24				Link and system level simulations added from GP-051062. Service continuity aspects and technical solutions added from GP-050791	0.2.0	0.3.0