Coverage Analysis of Skype VoIP Services over HSUPA

Renny E. Badra, Andrea Marcano and Hayat Abdulla

***Abstract*—Popular Voice-over-IP (VoIP) services such as Skype are frequently being accessed over cellular networks using a variety of radio technologies, of which High Speed Packet Data (HSPA) is one of the most commonly used by smartphones and other mobile devices. HSPA's uplink (HSUPA) is based on a highly flexible air interface with many settable parameters. This work analyzes HSUPA from the point of view of VoIP in order to formulate recommendations aimed at maximizing voice coverage while maintaining a quality of service that is comparable to that of standard circuit-switched (CS) cellular voice services. Moreover, it quantifies coverage for a number of vocoders used by Skype and compares it with conventional CS voice service provided over 3G cellular networks. Results show that a moderate penalty in cell coverage probability must be paid when switching from CS voice to Skype VoIP over HSUPA, even as its radio parameters are adjusted in order to maximize coverage.**

Index terms; cellular coverage; HSUPA; Skype; VoIP.

I. Introduction

With over one billion subscribers and 300 million active users worldwide, Skype is the most popular commercial application for peer-to-peer internet-based telephony and video telephony [1]. It supports VoIP calls between two Skype clients (E2E, end-to-end calls), and it also supports calls between a Skype client and a conventional land or mobile telephone line (E2O, SkypeOut calls). As digital cellular networks have spread to cover markets all over the world, more cellular subscribers are using Skype for voice calls from their data-enabled cellular devices, given the lower fees and increased connectivity associated to Skype.[·]

Third Generation (3G) cellular networks are commonly designed and optimized for providing circuit-switched (CS) voice services over its Wideband Code Division Multiple Access (WCDMA, also known as Release 99) air interface. Starting in 2005, High Speed Packet Access (HSPA, also known as 3.5G) systems have been overlayed on WCDMA cellular networks in order to provide faster bit rates and more capacity for nomadic data access. The HSPA system is exclusively designed for transmission of packet data supporting a wide range of Quality of Service (QoS) parameters, and consequently, it features a highly flexible and versatile radio interface in which many parameters can be independently adjusted.

Despite the increasingly mainstream presence of faster 4G cellular networks, 3G systems (WCDMA+HSPA) still represents a far larger share of the broadband market, with 1.8 billion subscriber worlwide (March 2015) compared to 497 million 4G subscribers [2], and is expected to remain dominant for the next few years.

In order to provide reliable VoIP services over HSPA with a QoS that is comparable to that of conventional CS voice services, several issues have been identified [3], among which uplink coverage is one of the most challenging [4]. Some of the questions surrounding the problem of maximizing coverage for VoIP over HSUPA (HSPA's uplink) are related to the well-known bit rate – coverage trade-off that is inherent to any wireless system; some other are impacted by the delay budget available for real-time voice services, and some other are tied to the limitations of the power amplifier at the User Equipment (UE) unit. As in any CDMA cellular system, capacity is also directly affected by choices aimed at improving coverage.

As a result of these considerations, it becomes clear the need for a study that analyzes the multiple parameters of HSUPA's air interface under the constrains posed by an actual VoIP application such as Skype. This work is aimed at achieving this goal from a multi-layer perspective that emphasizes traffic and RF issues. As an additional result, coverage for a number of vocoders used by Skype is quantified and compared with conventional CS voice service provided over WCDMA cellular networks.

In this article, Section II analyses the features of the Skype traffic that are relevant to this work. Section III describes the general methodology and the HSUPA and WCDMA uplink budget analysis that supports the study. Section IV argues in favor of the choice of parameters adopted in order to maximize VoIP coverage over HSUPA. Finally, results and conclusions are presented in Sections V and VI, respectively.

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Renny E. Badra and Hayat Abdulla are with Dept. of Electronics and Circuits, Universidad Simón Bolívar, Caracas, Venezuela. Email: renny@usb.ve. Phone: (+58)212-906-3669.

Andrea Marcano was with Dept. of Electronics and Circuits, Universidad Simón Bolívar, Caracas, Venezuela. She is now in with the Dept. of Photonics Eng,, Technical University of Denmark, 2800 Kongens Lyngby Denmark.

II. SKYPE TRAFFIC OVER HSUPA

Skype is a proprietary, closed-source, peer-to-peer internet application [5]. On the transport layer, Skype may operate on RTP/UDP or TCP, although RTP/UDP is widely preferred to carry Skype VoIP traffic.

Skype employs one of several voice codecs. For VoIP E2O calls, the G.729 codec is preferred. For Skype clients prior to version 4.0, the Sinusoidal Voice Over Packet Coder (SVOPC) [6] is the most commonly used codec for E2E calls, while starting with Skype 4.0, the more advanced SILK codec [7] has become available. G.729 provide a Constant Bit Rate (CBR) output, while the other two are of the Variable Bit Rate (VBR) type. In addition to being VBR, SILK also offers four different speech sampling rates: 4, 12, 16 and 24 kHz. Table I shows some of the parameters of these codecs that are relevant to this research, with emphasis on the 8- and 16-kHz SILK codecs, also known as the Narrowband (NB) and Wideband (WB) modes.

TABLE I. SKYPE VOCODERS AND ASSOCIATED TRAFFIC

CODEC	G.729	SVOPC	NB SILK	WB SILK
CBR/VBR	CBR	VBR	VBR	VBR
Sampling Rate (KHz)	8 kHz	16 kHz	8 kHz	16 kHz
Codec Frame interval (ms)	10	Variable 20 – 60	Variable 20 – 60	Variable 20 – 60
Codec output bit rate (kbps)	8	Variable 20 - 50	Variable 6 - 20	Variable 8 - 30
Resulting peak HSUPA MAC bit rate (kbps)	21.6	55.2 – 122.4	55.2	57.0 – 72.0

A key parameter for determining the characteristics of Skype traffic over HSUPA is the inter-frame interval. As seen in Table I, both VBR codecs feature variable frame intervals. However, it has been experimentally determined that, in steady-state, almost all inter-frame times take on a single value, which has been observed to be 20 ms for the SVOPC codec [8], and 20 and 60 ms for the SILK codec, when operating at its maximum and minimum bit rates, respectively [9].

A useful model for the Skype data sending rate $r_s(t)$, that is, the bit rate at the output of a Skype transmitting client, is [10]

$$r_s(t) = (1 - l(t)) \bullet (1 + r(t)) \bullet L_i(t), \qquad (1)$$

where $l(t)$ is the Skype-estimated packet loss rate, $r(t)$ is the redundancy ratio and $L_i(t)$ is the voice codec bit rate corresponding to its i-th operating mode. The output bit rate of both the SILK and the SVOPC codecs is tuned based on local information and network performance parameters such as packet loss ratio, jitter and round-trip time. As the codec bit rate is reduced due to networking constraints, so is reduced the quality of the reconstructed speech signal. On the other hand, the redundancy ratio quantifies the fraction of VoIP packets that are transmitted twice, and is adjusted by Skype between 0 and 1 as a function of the detected packet loss ratio. This work assumes a Block Error Rate (BLER) of 1% over the air interface, resulting in a Skype redundancy ratio r(t) of zero (no redundancy) [8], which in turn provides for a Skype sending rate that is approximately equal to the vocoder output rate. On the other hand, allowing for a higher BLER over HSUPA would cause $r(t)$ to increase, causing excessively high data rates over the air interface and thus significantly shrinking cell coverage [11].

In order to obtain the peak HSUPA E-DCH (Enhanced Dedicated Channel) data rate for each of the codecs, as shown in Table I, overhead bits from various protocols above (and including) MAC layer must be taken into consideration, such as RTP/UDP, IP, PDCP and RLC [12] (see Fig. 1). Header compression for RTP/UDP/IP headers using the RoHC algorithm is assumed [13]. Once all headers have been added to each VoIP packet, the MAC data rate displayed at the bottom of Table I is obtained. A TTI (Transmission Time Interval) value of 10 ms is assumed, since the lower TTI value of 2 ms would result in much higher peak rates, and consequently, severely reduced coverage [4]. Data rates displayed represent a peak value, that is, they are the instantaneous MAC rates whenever a VoIP packet is being transmitted (transmission over HSUPA is not continuous). Note that for the NB SILK codec, the maximum and the minimum MAC bit rates coincide, as a consequence of the fact that the higher bit rates in SILK are mostly due to a reduction of the inter-frame time rather than an increase in packet size.

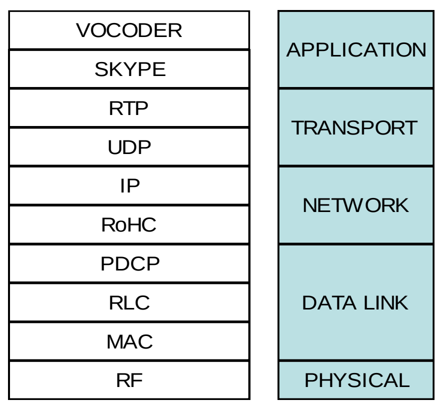

Fig. 1. Assumed protocol stack for Skype VoIP over HSUPA.

In order to be able to perform a benchmark comparison with conventional circuit-switched voice services offered over WCDMA (3G) cellular networks, two modes of both NB and WB AMR (Adaptive Multi Rate) codecs have been selected, which have a MOS (Mean Opinion Score) performance that is similar to the corresponding NB and WB Skype vocoders. Table II summarizes the relevant parameters of these codecs. The resulting WCDMA DPDCH (Dedicated Physical Data Channel) data rates at MAC level shown at the bottom of Table II take into consideration all of the relevant protocol overhead bits, and also includes the effect a 3.4-kbps concurrent signaling channel. As opposed to HSUPA, these data rates are continuously present over the WCDMA uplink, since the radio connection is circuit-switched. The circuit-switched AMR benchmark voice service is typically set to achieve BLER=1% in order to provide acceptable MOS levels.

TABLE II. WCDMA VOCODERS SELECTED AS REFERENCE AND THEIR ASSOCIATED TRAFFIC

CODEC	NB AMR	WB AMR
CBR/VBR	CBR	CBR
Sampling Frequency	8 kHz	16 kHz
Codec Frame interval	20 ms	20 ms
Codec output bit rate	12.2 kbps	15.85 kbps
Resulting WCDMA MAC bit rate	18.3 kbps	21.85 kbps

III. METHODOLOGY

A. Simulation Set-up and Objective

In order to estimate the coverage probability of each of the services specified in Section II, a set of numeric simulations of the uplink power budget has been performed and evaluated via Monte Carlo trials. Macro cells are assumed to be hexagonal, with the cell radius as one of the controlled parameters. The number of Monte Carlo simulation points within each cell, each corresponding to a different random UE location, was set to 100,000, of which 1/3 corresponds to an outdoors, street-level mobile user location and 2/3 corresponds to indoor users.

Coverage probability, the key output of the simulations, is defined as the fraction of Monte Carlo trials corresponding to locations inside the cell in which the minimum transmit power required to close the uplink P_{min} is such that

$$P_{\min} \le P_{UE} \quad , \tag{2}$$

where P_{UE} is the UE maximum nominal transmit power.

Link budget analysis is used to determine the P_{min} value that closes the uplink. The link budget equation expressed in dBm and dB is

$$P_{\min} = R_{sen} + L_p + \sum_i L_i - \sum_i G_i \quad , \tag{3}$$

where R_{sen} is the receiver sensitivity, L_p is the propagation loss including loss margins, L_i is the i-th system loss factor and G_i is the i-th system gain factor. Receiver sensitivity is calculated through

$$R_{sen} = -174\frac{\text{dBm}}{\text{Hz}} + 10\log(R_{b,MAC}) + NF + \left(\frac{E_b}{N_t}\right)_{comb} , \tag{4}$$

where $R_{b,MAC}$ is the service MAC bit rate in bits per second (as in Tables I and II for each of the vocoders under study), NF is the base station receiver noise figure in dB, and $(Eb/Nt)_{comb}$ is the minimum requirement for the "combined" MAC-level bit energy to noise ratio, also in dB (see next section).

B. Combined E_b/N_t

The "combined" E_b/N_t is defined as the average bit energy to noise-plus interference required to achieve a certain performance on the HSUPA E-DCH. The term "combined" accounts for the fact that this E_b/N_t value must reflect the overhead power that needs to be allocated to other uplink code channels. Different combinations of uplink code channels are available for HSUPA; in this work we have chosen to consider, in addition to the E-DPDCH (Enhanced Dedicated Physical Data Channel, which carries the E-DCH), one of each of the following physical code channels: DPCCH (Dedicated Physical Control Channel), DPDCH (Dedicated Physical Data Channel), HS-DPCCH (High Speed DPCCH) and E-DPCCH (Enhanced DPCCH). The first two of these code channels (DPCCH and DPDCH) are Release 99 legacy channels used to carry PHY control signals (including the pilot channel) and L3 signaling messages, respectively. The HS-DPCCH supports the use of the HS-DSCH (High Speed Downlink Shared Channel) for concurrent HSDPA/HSUPA operation and the E-DPCCH carries MAC/PHY layer control information for the E-DCH.

The combined Eb/Nt requirement for the E-DCH can be calculated using [14]

$$\left(\frac{E_b}{N_t}\right)_{comb} = G_p\left(\frac{E_c}{N_t}\right)_c \left[1 + \left(\frac{\beta_{ed}}{\beta_c}\right)^2 + \left(\frac{\beta_{ec}}{\beta_c}\right)^2 + \left(\frac{\beta_d}{\beta_c}\right)^2 + \left(\frac{\beta_{hs}}{\beta_c}\right)^2\right], \tag{5}$$

where $(E_c/N_t)_c$ is the chip energy to noise plus interference required for the pilot (DPCCH) channel, and β_c, β_d, β_{ec}, β_{ed}, β_{hs} are the relative amplitude values given to the DPCCH, DPDCH, E-DPCCH, DPDCH and HS-DPCCH in the composite CDMA uplink signal. On the other hand, the processing gain G_p is defined as

$$G_p = \frac{W}{R_{b,MAC}} \quad , \tag{6}$$

where W is the signal noise-equivalent bandwidth (approximately equal to the chip rate).

Simulation results for $(E_c/N_t)_c$ have been reported for different HSUPA configurations [14], depending on various air interface settings such as the TTI, the TBS (Transmission Block Size), the required BLER and the number of HARQ (Hybrid ARQ) retransmissions. However, these $(E_c/N_t)_c$ values tend to hover around the -21 to -22 dB range, showing little variation, as expected. In this work, as explained in Sect. II, a TTI value of 10 ms and a BLER value of 1% have been chosen. Due to the delay budget constrains for voice services, the number of allowed HARQ retransmissions has been set to one, in order to keep the maximum HSUPA delay under 75 ms. Finally, the values of the TBS depend on the IP packet sizes: 186 bits for G.729, 522 to 1194 bits for SVOPC, 522 bits for NB SILK and 540 to 690 bits for WB SILK. The combined E_b/N_t requirement based on eq. (5) and $(E_c/N_t)_c$ values obtained via simulations over the 3GPP Pedestrian A 3 kph (PA3) channel model, and assuming a Category 5 UE [14] are given in Table III, which also displays the combined E_b/N_t values for the reference WCDMA services (NB/WB AMR) [11].

TABLE III. REFERENCE VOCODERS AND ASSOCIATED TRAFFIC

Vocoder	System	$R_{b,MAC}$	$(E_b/N_t)_{comb}$
G.729	HSUPA	21.6 kbps	7.8 dB
SVOPC		55.2 kbps	4.7 dB
		122.4 kbps	2.7 dB
NB SILK		55.2 kbps	4.7 dB
WB SILK		57.0 kbps	4.7 dB
		72.0 kbps	4.0 dB
NB AMR	WCDMA	18.3 kbps	4.5 dB
WB AMR		21.55 kbps	4.3 dB

Relative amplitude coefficients (β factors) were set as a function of the channel configuration and TBS, according to recommendations [14, 15]. In particular, the $(\beta_{ed}/\beta_c)^2$ ratio, also known as T/P (traffic-to pilot ratio), takes on values ranging from 1 dB (for the smallest TBS of 186 bits), to 3 dB (for a medium-sized TBS of 540 bits), up to 7 dB (for the largest TBS of 1194 bits).

C. Maximum Power Reduction (MPR)

The multi-code operation of HSUPA causes time variations in the envelope of the composite CDMA signal, which in turn leads to large Peak-to-Average Power ratios (PAPR). It is well known that high PAPR signals put significant stress on the power amplifier (PA) of the UE, which is required to provide a large linear operating range in order to reduce non-linear distortion and thus keep adjacent channel interference under the specified limits. In order to guarantee linear PA operation, a service-specific power back-off must be introduced, which is in effect as a reduction in maximum transmit power (MPR) and translates into the uplink budget as a signal loss in eq. (3).

Envelope variations, and consequently MPR, clearly depend on the combination of uplink code channels and their relative amplitudes (β factors defined above), as well as on the spreading factors used in each of these code channels. Since the code channel amplitudes depend on the data rate, a link between the TBS and the MPR has been established [15]. The reason is simple: at higher data rates and TBS values, the relative power of the E-DCH with respect to the rest of the code channels grows, thus making envelope variations less significant and lowering MPR. Unfortunately, VoIP services use relatively small packet sizes, which increases MPR. For TBS values such as those given in Sect. IIIb, MPR has been reported to take on values ranging from 2.4 to 2.5 dB [14].

D. Other Link Budget Parameters

Excluding propagation losses, the other system loss factors L_i considered in (3) are the cell load factor (a measure of co-channel interference, [16]) and the base station (BS) cable losses. The gain factors G_i considered in (3) are the base station antenna gain and the soft handover gain, which is defined as the amount of transmit power that the mobile station is able to save due to soft handover in progress [16]. The full list of link budget parameters used in (3), as employed in the simulation parameters (excluding propagation, see next Section), are given in Table IV. These parameters apply to simulations involving both WCDMA and HSUPA systems.

TABLE IV. RF PARAMETERS USED IN SIMULATIONS

Parameter	Value
UE max transmit power	23 dBm (1900 MHz)
UE ant. gain + cable loss	0 dB
Receiver Noise Figure	5.0 dB
Receiver Sensitivity	Service-dependent (see Eq. (4))
BS Antenna Type	Sectorized, 90° beamwidth
BS Antenna Gain	17 dBi
BS Cable losses	2 dB
Uplink Load Factor	3.0 dB (50% cell load)
BS ant. diversity gain	Included in $(Eb/Nt)_{comb}$

E. Urban Cellular Propagation Model

The uplink cellular propagation model employed accounts for signal power loss incurred between the transmit and receive antennas, including loss margins. It distinguishes between outdoors and indoors locations. In both scenarios, the model is comprised of five terms,

$$L_P = L_D + L_{Sh} + L_S + L_{Pn} + L_B \quad , \tag{7}$$

where L_D is the distance-dependent propagation loss, L_{Sh} is the shadowing factor, L_S is the small-scale fading component, L_{Pn} is the building penetration loss and L_B is the power loss due to interaction with the user's body, all in dB.

The distance-dependent propagation loss L_D for outdoors locations considers only non-line-of-sight (NLOS) locations, since line-of-sight (LOS) locations are highly unlikely [17]. Following 3GPP recommendations [17], the Hata/COST231 model is used for NLOS. The shadowing factor L_{Sh} accounts for large-scale variations of signal strength. It is accurately described by a random variable (RV) with a zero-mean log-normal probability distribution, that is, the distribution of the RV expressed in dB is normal. In this study, the standard deviation of such RV has been set to 8 dB (outdoors NLOS locations) and 11 dB (indoors locations), following recommendations [17]. Indoor and outdoor probabilities are set to, respectively, 2/3 and 1/3

The small-scale fading component L_S models signal variations over short distances. The amplitude of this fading factor for NLOS propagation is modeled through a Rayleigh RV. The average building penetration power loss for indoor locations has been reported to be around 10 dB, with additional losses as a function of the distance to the nearest window and to the ground [18]. Accordingly, in this work the building penetration loss has been assumed to be a RV with a 12 dB mean and a 3 dB standard deviation.

For outdoors locations there are no penetration losses. Finally, the RF power loss caused by RF interaction with the human body of the user L_B is set to a fixed value of 3 dB [16]. Table V summarizes the propagation models and parameters adopted.

TABLE V. PROPAGATION MODELS AND PARAMETERS USED IN SIMULATIONS

Location	Outdoors	Indoors
Environment	Urban	
Frequency	1900 MHz (UMTS band)	
UE location	Street-level pedestrian	In-building
UE location probability	1/3	2/3
LOS/NLOS	NLOS at all locations	
Distance-dependent Propag. Loss	Hata / COST 231	
Shadowing Std. Dev.	8 dB	11 dB
Small-Scale Fading	Rayleigh	
Building Penetration Loss	0 dB	Mean: 12 dB St. Dev: 3 dB
Body Loss	3 dB	

F. Cell Size

Three urban macrocell sizes have been considered, each corresponding to a different coverage probability for the benchmark 12.2-kbps AMR CS voice service over WCDMA. Table VI summarizes the three simulation scenarios, and the coverage probability obtained for the 12.2-kbps circuit-switched AMR speech service, which is taken as a reference.

TABLE VI. CELL SIZE SCENARIOS

Simulation Scenario	Larger Cells	Medium-sized Cells	Smaller Cells
Cell radius	1540 m	1225 m	880 m
Cell-area coverage probability for 12.2-kbps CS AMR speech	85 %	90 %	95 %

IV. SUMMARY OF IDENTIFIED SYSTEM TRADE-OFFS

As outlined in the previous section, the coverage study in this paper assumes that a number of HSUPA parameters have been properly adjusted in order to offer Skype VoIP within the general boundaries of a QoS that is comparable to that offered by CS voice services over WCDMA, while maximizing cell coverage. Among these parameters, the following have been found to be of paramount importance:

- BLER, which must be adjusted to 1 or 2% in order to prevent the Skype traffic from shooting up excessively due to frame duplication at the application layer. Allowing for a higher BLER would certainly reduce the $(E_b/N_t)_{\mathrm{comb}}$ requirement at the expense of significantly increased packet size (up to twofold), and therefore increased TBS and data rate, which would seriously reduce coverage and also impact cell capacity.
- TTI, which has been set to 10 ms in order to keep the instantaneous transmit rates in HSUPA as low as possible. Setting TTI to 2 ms would significantly reduce delay but would also cause data rates to increase and cell coverage to shrink. Setting TTI to 10 mS also achieves an unexpected benefit, which is to better spread out in time the multiple access interference, which in a non-orthogonal CDMA uplink such as HSUPA's tends to smooth variations in cell load, and in general, stabilize the performance of the uplink.
- TBS, which has been set to the lowest possible value that can accommodate one single VoIP packet with RTP/UDP/IP header compression. Allowing for larger TBS values (by aggregating consecutive VoIP packets, or by not compressing RTP/UDP/IP headers) would also negatively impact coverage, and possibly delay. Larger TBS values also reduce cell capacity.
- The choice of TBS determines the best combination for the relative amplitude parameters (β factors). By minimizing TBS, the β factors are such that MPR is certainly maximized, which is viewed as an inevitable price to be paid in order to maximize coverage.
- Number of HARQ retransmissions, which has been limited to one (two transmissions in total) in order to keep one-way VoIP packet delay under 75 mS. Setting the number of retransmissions to its maximum value of two (three transmissions in total) would somewhat reduce the $(E_b/N_t)_{\mathrm{comb}}$ requirement at the expense of a transmission delay that can be as high as 120 ms, and much more jittery packets.

V. SIMULATION RESULTS AND ANALYSIS

Table VII summarizes the simulation results, that is, the probability of cell-area coverage as a function of cell size. In the case of VBR vocoders (SVOPC and SILK) the two displayed values correspond to best and worst coverage probability, which are aligned with the maximum and minimum vocoder output rate (except for NB SILK, which as explained in Sect. II achieves the same HSUPA transmission rate for both maximum and minimum vocoder output rate).

TABLE VII. SIMULATION RESULTS: PROBABILITY OF CELL-AREA COVERAGE

Vocoder	**Technology**	**Larger Cells**	**Medium-sized Cells**	**Smaller Cells**
NB AMR	CS WCDMA	85%	90%	95%
G.729	Skype-HSUPA	72%	79%	88%
NB SILK	Skype-HSUPA	70%	77%	86%
WB AMR	CS WCDMA	84%	89%	94%
SVOPC	Skype-HSUPA	66-70%	74-77%	84-86%
WB SILK	Skype-HSUPA	68-69%	76-77%	85-86%

As expected, narrowband vocoders offer slightly better coverage probability than wideband vocoders. However, when comparing CS WCDMA coverage (NB or WB) with the corresponding NB/WB VoIP coverage, it is possible to appreciate a significant loss in probability of coverage: 13-15% for NB vocoders, 14-18% for WB vocoders (larger cells); 9-13% for NB vocoders, 8-10% for WB vocoders smaller cells).

Among NB Skype vocoders, G.729 does offer better coverage performance than NB SILK, but only by about 2%. As opposed to previous findings reported for Skype use over WCDMA [11], Skype's CBR vocoder G.729 does not offer significantly better coverage than NB SILK.

On the other hand, among WB Skype vocoders, SILK offers a slightly better coverage probability than SVOPC; moreover, the differences between maximum-rate and minimum-rate coverage probability are higher for SVOPC (2 to 4%) than for WB SILK (barely 1%).

VI. CONCLUSIONS

Firstly, it can be concluded that VoIP coverage using Skype VoIP services over HSUPA is always inferior to that offered by equivalent CS voice services over WCDMA. The penalty in coverage for switching to Skype VoIP ranges from 13 to 18% of cell area in the case of larger cells, and from 8 to 13% of cell area in the case of smaller cells. As expected, coverage for WB voice services is slightly worse than for NB voice.

Secondly, it has been also found that the changes in vocoder mode (and therefore, bit rate) in the newer Skype vocoder SILK do not translate into significant changes in the probability of coverage, because of the fact that the higher bit rates in SILK are mostly due to a reduction of the inter-frame interval rather than an increase in the packet size. As a matter of fact, it has been found that in the case of NB SILK there are no differences in cell coverage between the maximum and the minimum vocoder output rates.

Finally and more significantly, this work has confirmed and expanded on previous findings related to the trade-off situations that arise when optimizing HSUPA's air interface for VoIP services, in this case particularly applied to a popular commercial application such as Skype. The list of key parameters that have been found in achieving such a goal are BLER, TTI, TBS and the number of HARQ retransmissions, for which a set of recommendations has been formulated.

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