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A White Paper by the NGMN Alliance

NGMN Radio Access Performance Evaluation Methodology

next generation mobile networks

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Next Generation Mobile Networks Radio Access Performance Evaluation Methodology

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Abstract

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CONTENT

INTRODUCTION

The NGMN project is an initiative by a group of leading operators to provide a vision for technology beyond 3G for the competitive delivery of broadband wireless services to increase further end-user benefits. In the White Paper 3.0 [1], recommendations for common system features, radio access network, core network, terminals and services are made, and operator requirements are established.

In this paper, an evaluation methodology for some of the NGMN requirements [1] is proposed. This includes a more detailed definition of the high-level metrics in [1], and a set of common evaluation scenarios. The objectives are to verify the performance of different NGMN system proposals under the same conditions and to align the evaluation parameters of different partners developing the same technology.

The metrics should reflect the user experience and operator requirements as closely as possible whilst being reasonably easy to be assessed by simulations and trials.

Different standardization bodies addressing systems within the scope of NGMN have developed their own set of metrics and evaluation scenarios, which makes a reasonably fair "apples with apples" comparison difficult. Also, these evaluation methodologies are not always completely defined or consistent leading to large variations of results even if different sources simulate the same technology with the same methodology.

3GPP LTE has developed a list of performance requirements in [3]. Recently, operators made a proposal for a physical layer framework for evaluation of LTE [4].

Evaluation methodologies for wide area systems were developed also in 3GPP2 (e.g. [6]) and IEEE 802.20 [9]. The recently formed IEEE 802.16m group discusses also an evaluation methodology, as does IEEE 802.16j [4].

This paper reuses metrics and scenarios from these mentioned standardization initiatives wherever appropriate.

Being technology-agnostic, this paper specifies only metrics and scenarios, but not the elements of the technology itself (e.g. which modulation and coding scheme). However, any evaluation of a system according to the NGMN methodology shall provide an exact description of the underlying technology elements.

NGMN is an alliance by a group of leading mobile operators, industrial partners, and academic advisors to provide a vision for technology evolution beyond 3G for the competitive delivery of broadband wireless services. The vision of the NGMN alliance is that of a global mobile society where any service can be accessed through a personal device connected via a wireless mobile network. The key objective of the alliance is to create a virtuous cycle of investment, innovation, and adoption of mobile broadband services with competitive price-performance ratios.

1 EVALUATION METRICS

The NGMN White Paper 3.0 [1] mentions the following metrics:

- Uplink and Downlink data rates for a network as a whole assuming all cells are interference limited
- Spectrum efficiency
- Core latency
- RAN latency
- E2E latency
- Mobility support with service continuity through a minimum of 120 km/hr

[1] also gives essential and preferred recommendations for them.

1.1 FAIRNESS

From an operator view, fairness is very important since users expect to have the same experience regardless whether they are close to a base station or at the cell edge. Average throughput and spectral efficiency figures make only sense if a fairness criterion is fulfilled.

Figure 1 shows the cumulative distribution function (cdf) of the normalized user throughput, i.e. the mean user throughput has the value 1.0. Different schedulers have different fairness behaviour. The normalized throughput bound is selected as fairness criterion. Alternative fairness criteria are fairness index or satisfied user criterion.

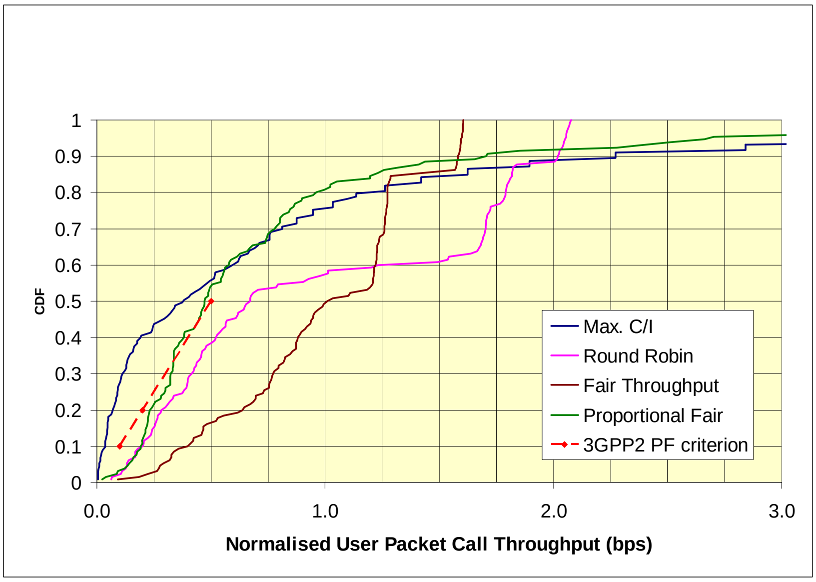

Figure 1 Cumulative Distribution Function (CDF) of Normalized User packet Call Throughput: Example

Metric *The **Normalized Throughput Bound** defines a region in the throughput cdf plot, where the throughput should at least be right of it. This region is defined by the line given by the three points in Table 1.*

The interpretation is that at least 90% of the users should have at least 10% of the average user throughput. This methodology is used in 3GPP2 [8], IEEE 802.16j [7] and IEEE 802.20 [9].

Normalized Throughput w.r.t. average user throughput	cdf
0.1	0.1
0.2	0.2
0.5	0.5

Table 1 Normalized Throughput Bound

1.2 DATA RATES AND SPECTRAL EFFICIENCY

1.2.1 PEAK USER DATA RATE

The peak user data rates und user throughput metrics shall consider all overhead (control channels, pilots, guard interval, coding rates etc)

Metric *The **peak user downlink data rate***

Metric *The **peak user uplink data rate***

These metrics shall be determined by system analysis.

1.2.2 THROUGHPUT

The peak user data rate is a theoretical number since the aggregate cell throughput is in most circumstances shared by all users within a cell. The throughput figures should reflect the user experience, no matter where the user is placed in the cell. Therefore the cell-edge user throughput, defined as a cdf operating point, is the relevant figure of merit. The throughput is a function of the base-station separation and number of users per site. For throughput calculation, the fairness criterion and the delay criterion shall be fulfilled. The users are placed randomly (uniform distribution) in the cell area. For throughput figures, a fully loaded multi-cell system has to be assumed. One possible simulation methodology is cell-wrap-around.

The throughput figures should be obtained for a fully functioning system, i.e. with uplink and downlink connections established to the users and successful receiver processing of the necessary control, synchronization and pilot channels.

Metric *The **full-buffer downlink user throughput cdf** as a function of number of users and site-site distance for a given fairness and delay criterion in a fully loaded network with a full buffer assumption.*

Metric *The **full-buffer uplink user throughput cdf** as a function of number of users and site-site distance for a given fairness and delay criterion in a fully loaded network with a full buffer assumption.*

Metric *The **service-mix downlink user throughput cdf** as a function of number of users and site-site distance for a given fairness and delay criterion in a fully loaded network with a service mix assumption.*

Metric *The **service-mix uplink user throughput cdf** as a function of number of users and site-site distance for a given fairness and delay criterion in a fully loaded network with a service mix assumption.*

Metric *From these cdfs, **the average and 95%ile user throughputs for uplink, downlink, full-buffer and mixed service** shall be calculated, respectively.*

Metric *The **aggregate downlink full-buffer site throughput** is the number of successfully transmitted information bits per second that a site can serve for a given number of users and site-site distance for a given fairness and delay criterion in a fully loaded network.*

Metric *The **aggregate uplink full-buffer site throughput** is the number of successfully transmitted information bits per second that a site can serve for a given number of users and site-site distance for a given fairness and delay criterion in a fully loaded network.*

Metric *The **aggregate downlink service-mix site throughput** is the number of successfully transmitted information bits per second that a site can serve for a specific service traffic mix and site-site distance for a given fairness and delay criterion in a fully loaded network. The same scheduler shall be used for all measurements.*

Metric *The **aggregate uplink service-mix site throughput** is the number of successfully transmitted information bits per second that a site can serve for a specific service traffic mix and site-site distance for a given fairness and delay criterion in a fully loaded network. The same scheduler shall be used for all measurements.*

1.2.3 SPECTRAL EFFICIENCY

The spectral efficiency is an important criterion for operators who have acquired spectrum licenses for substantial amounts of money, and they strive to use the spectrum as efficiently as possible.

The spectral efficiency shall be given per cell (equals sector) for a reference three fold sectorization. The spectrum bandwidth used for spectral efficiency calculations shall be the spectrum block assignment size.

For spectral efficiency calculations, the fairness criterion shall be fulfilled. Spectral efficiency figures should be given for different site-site distances.

Metric *The **downlink spectral efficiency** (in bits/s/Hz/cell) is the aggregate full-buffer user or traffic-mix throughput per spectrum block assignment bandwidth. The same satisfied user, fairness, delay and outage criteria as for the aggregate throughput shall be used.*

Metric *The **uplink spectral efficiency** (in bits/s/Hz/cell) is the aggregate full-buffer user or traffic-mix throughput per spectrum block assignment bandwidth. The same satisfied user, fairness, delay and outage criteria as for the aggregate throughput shall be used.*

1.2.4 VOIP CAPACITY

The VoIP capacity shall be estimated using the traffic model and satisfied user criteria as defined in Annex B.

For VoIP capacity evaluation, an appropriate HARQ retransmission rate of the first transmission shall be used taking into account the allowable delay.

Metric *The **VoIP capacity** is the maximum number of satisfied users supported per site in downlink and uplink.*

1.2.5 BROADCAST AND MULTICAST SERVICE EVALUATION

In [1], the support of broadcast and multicast services is demanded, and optimized MBMS is not precluded. In [4], a system evaluation methodology for SFN(Single Frequency Network) transmission of an MBMS service on a dedicated carrier is given.

Metric ***System throughput of broadcast transmission*** *is calculated assuming that 95% of locations in a nominal coverage area experience a PER of 1% or less.*

Metric ***Spectral efficiency of broadcast transmission*** *is the system throughput of broadcast transmission normalized by the occupied bandwidth*

Metric ***The maximum broadcast inter site distance (ISD)*** *is obtained assuming a system throughput of 1 bit/s//Hz in 95% of locations in a nominal coverage area with a PER of 1% or less.*

1.2.6 LATENCY

A sufficiently low latency is important for the user experience, especially for some envisaged service classes. For the performance of certain applications, like the TCP protocol, the delay jitter (variation of the delay) is important.

The latency can be obtained by two approaches: system analysis and simulations. System analysis makes reasonable assumptions (e.g. HARQ retransmission probability) of the point of operation and is therefore reasonably straightforward. Simulations of latency measures have to consider a lot of different system building blocks and are therefore more challenging. However, simulation-based statistics of the delay provide a higher confidence level with respect the results matching to the user experience in a real system.

1.2.7 CONNECTION SETUP LATENCY

For interactive applications, the response latency is very important, even if just very small packets are transmitted. The radio access system might have different connection states and transition mechanisms. The connection setup latency (also first packet latency or C-plane latency) contains the transition time from one state to another plus the time taken between the first data packet successfully reaching the receiver reference point.

Examples for states are:

- “OFF” (IDLE, switched off, NULL)
- “IDLE” (STANDBY, SLEEP, Listening)
- “ACTIVE” (READY, ON, Data Transfer)
- “STANDBY” (HOLD)

The first packet latency should be given for any state transition within the system. For analysis, the procedure for random access shall be used as defined in the system, and the delay, retransmissions, and the L1/L2/L3 processing delays shall be considered. For HARQ, typical retransmission rates of the first transmission (e.g. 20%) shall be used.

Metric — *The **connection setup latency** is calculated for all relevant state transitions defined within a system with reasonable assumptions using system analysis.*

Metric — *The **cdf of the connection setup latency** is calculated for all relevant state transitions defined within a system for arbitrarily placed users in a fully loaded cell based on simulations.*

Metric — *The **guaranteed connection setup latency** is the 95-ile of the cdf of the packet latency calculated for all relevant state transitions defined within a system based on simulations.*

1.2.8 RADIO ACCESS TRANSMISSION LATENCY

The radio access transmission latency is defined in terms of the one-way transit time between a packet being available at the IP layer (Tx reference point) in either the UE / Radio Access Network and the availability of this packet at IP layer (Rx reference point) in the Radio Access Network / UE.

Note: the location of the reference points depends on radio protocol architecture (e.g. for LTE reference points are on top of PDCP in UE and on top of PDCP in E-UTRAN). Figure 2 illustrates the radio access transmission latency.

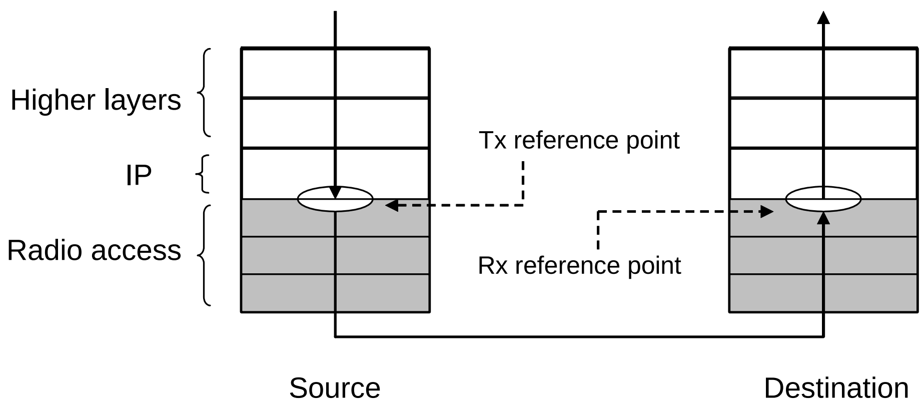

Figure 2 Illustration of reference points for Radio Access Transmission Latency

For system analysis, typical HARQ retransmission rates of the first transmission (20%) should be included, and simulations shall be made in a multi-cellular system under full load, with users distributed evenly within the cells.

The radio access transmission latency needs to be defined for 2 cases. In the pre-scheduled case, packets are received at the transmitter reference point with packet transmission already anticipated with resources already allocated. Examples of pre-scheduled bearers the persistent scheduling of VoIP calls in 3GPP LTE, or a UGC fixed bandwidth WiMAX bearer. The unscheduled case is applicable for a non real time bearer where additional delay is incurred to allow the scheduler to allocate resource, and in the case of the uplink, for the terminal to request resource for the packet transmission. Uplink and downlink latencies can differ and so are differentiated with different metrics.

Metric *The **one way downlink radio access network pre-scheduled transmission latency** based on system analysis with realistic assumptions*

Metric *The one **way uplink radio access network pre-scheduled transmission latency** based on system analysis with realistic assumptions*

Metric *The one **way downlink radio access network unscheduled transmission latency** based on system analysis with realistic assumptions (32 byte packet)*

Metric *The **one way uplink radio access network unscheduled transmission latency** based on system analysis with realistic assumptions (32 byte packet)*

Metric *The **cdf of the two-way radio access pre-scheduled network transmission latency**.*

Metric *The **cdf of the two-way radio access unscheduled network transmission latency**.*

Metric *The **simulation-based two-way pre-scheduled radio access radio access network transmission latency** is the 95%-ile of the cdf of the one-way access network transmission latency in a loaded network.*

Metric *The **simulation-based two-way radio access unscheduled radio access network transmission latency** is the 95%-ile of the cdf of the one-way access network transmission latency in a loaded network.*

Metric *The **radio access network jitter** is the average of the standard deviation of the one- radio access radio access network transmission latency of each user.*

Metric ***The one way core network packet transfer delay.** This is the time between reception of a packet by the core network and the sending of the packet from the other side of the core network.*

1.2.9 THE END-TO-END PACKET CALL LATENCY

A packet call is a sustained burst of user data (such as a web-page or a file).

Metric *The **cdf of the end-to-end packet call latency** is determined by measuring the time from the initial user request to completion at application level.*

Metric *The **end-to-end packet call latency** is the 95%-ile of the cdf of the one-way air-interface delay in a loaded network.*

1.2.10 HANDOVER INTERRUPTION TIME

The handover within one NGMN system or to another RAN (e.g. UTRAN or GERAN) can cause service interruption times and packet losses.

Metric *The **interruption time during a handover of real-time services within an NGMN system**.*

Metric *The **interruption time during a handover of real-time services between an NGMN system and a different RAN.***

Metric *The **interruption time during a handover of non-real-time services within an NGMN system.***

Metric *The **interruption time during a handover of non-real-time services between an NGMN system and a different RAN.***

The handover setup should be lossy, i.e. no additional mechanisms should be used.

1.3 OTHER METRICS

1.3.1 CONTOUR PLOT OF MINIMUM SERVICE LEVEL

Metric *The **contour plot of minimum service level** shows the 90%-ile of the number of full-buffer users which exceed a certain data rate for at least 80% of the time versus a sit-site separation in a fully loaded network.*

The load (number of users per site) versus the site separation (in kilometres) at a given minimum service level indicates the throughput/coverage trade-off. The minimum service level (Mbps) is exceeded for a given percentage of time (e.g. 80%) for a given outage (5% or 10%). Figure 3 shows an example contour of minimum service level from [9].

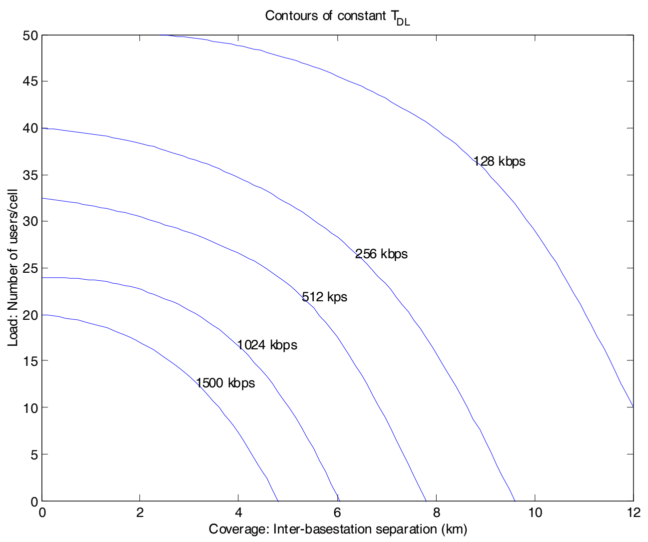

Figure 3 Example of Contours of Minimum Service Level. From [9].

1.3.2 USER GEOMETRIES AND INTERFERENCE STATISTICS

The geometry is the ratio of the strongest signal to the sum of the interfering signals and the noise. The full bandwidth should be considered and fast fading effects should not be modelled.

Metric *The **cdf of users's geometries shall be given for the downlink**.*

Metric *The **cdf of user's interference over thermal noise shall be given for the uplink**. This cdf depends on the power control scheme and thus also characterizes the power control algorithm used.*

1.3.3 NORMALIZED USER THROUGHPUT TO SNR MAPPING IN AWGN CHANNEL CONDITIONS

The Link-to-System interface can be calibrated by plotting the normalized user throughput in a given frequency bandwidth (in bits/s/Hz) vs. the signal-to-noise ratio (SNR), if an AWGN channel is assumed. Also, this metric can be replicated in laboratory trials and simulation results can be compared to measurement results.

Metric ***The table/graph of mapping of normalized user throughput to SNR in AWGN channel conditions should be given for the downlink for different antenna configurations.***

Metric ***The table/graph of mapping of normalized user throughput to SNR in AWGN channel conditions should be given for the uplink for different antenna configurations.***

This mapping should include the effects of coding and modulation schemes, spatial processing schemes, HARQ, pilot and control channel overhead, channel estimation, receiver structure and reflect what could be measured in a laboratory test or field trial. The normalized user throughput is the data rate of a single user within a given frequency bandwidth and is expressed in bits/s/Hz.

1.3.4 NORMALIZED USER THROUGHPUT TO GEOMETRY MAPPING IN REALISTIC CHANNEL CONDITIONS

The Normalized User Throughput to SNR Mapping in AWGN Channel Conditions has its limitations in describing the performance in real-world networks; it has to be extended therefore to more realistic channel conditions. The mapping of the normalized user throughput to geometry is a very useful measure for link budget calculations and coverage planning, even if this approach has limitations in accuracy.

Metric: ***The table/graph of mapping of normalized user throughput to geometry should be given for the downlink for different antenna configurations.***

Metric: ***The table/graph of mapping of normalized user throughput to geometry should be given for the uplink for different antenna configurations.***

This mapping should include the effects of coding and modulation schemes, spatial processing schemes, HARQ, pilot and control channel overhead, channel estimation, receiver structure and reflect what could be measured in a laboratory test or field trial. The normalized user throughput is the data rate of a single user within a given frequency bandwidth and is expressed in bits/s/Hz.

1.3.5 CONTROL CHANNEL AND PILOT OVERHEAD

Metric *The* ***control channel and pilot overhead*** *is the percentage of radio resources utilized for signalling, control channels and pilots. This value shall be obtained under the same conditions as used for spectral efficiency calculations, preferably by explicit modelling of the control channels usage in the system simulations. The overhead shall be split into individual items and be given for the uplink and downlink. Even for low speed simulations, the control channel configuration should be designed to be robust enough to support reliable cell edge operation of 120 km/h users.*

Table 2 shows an example how overhead elements could be split in an overhead budget. The appendix gives also an example of overhead budget for different technologies.

<table>
<tr><th></th><th colspan="2">Overhead</th><th>Method of Calculation</th></tr>
<tr><td rowspan="10">Major impact</td><td rowspan="3">Symbol structure</td><td>cyclic prefix,</td><td></td></tr>
<tr><td>guard band</td><td></td></tr>
<tr><td>UL/DL spacing</td><td></td></tr>
<tr><td rowspan="5">Reference signals(1/2/4 antennas)</td><td>DL Pilots</td><td></td></tr>
<tr><td>UL Pilots</td><td></td></tr>
<tr><td>Sounding</td><td></td></tr>
<tr><td>Synchronization</td><td></td></tr>
<tr><td>Preambles</td><td></td></tr>
<tr><td colspan="2">Control signaling</td><td></td></tr>
<tr><td colspan="2">DL SCH or Preamble</td><td></td></tr>
<tr><td rowspan="5">Should be analyzed</td><td colspan="2">System information broadcast</td><td></td></tr>
<tr><td colspan="2">HARQ ACK/NACKs</td><td></td></tr>
<tr><td colspan="2">Uplink feedback (CQI, MIMO feedback, BW requests)</td><td></td></tr>
<tr><td colspan="2">Timing advance</td><td></td></tr>
<tr><td colspan="2">Power control</td><td></td></tr>
<tr><td rowspan="6">Minor impact</td><td colspan="2">ARQ related signaling</td><td></td></tr>
<tr><td colspan="2">DRX related signaling</td><td></td></tr>
<tr><td colspan="2">HO / cell change related signalling</td><td></td></tr>
<tr><td colspan="2">Initial access</td><td></td></tr>
<tr><td colspan="2">Measurement reporting</td><td></td></tr>
<tr><td colspan="2">NAS signaling</td><td></td></tr>
</table>

Table 2: Example Overhead Budget

2 EVALUATION SCENARIOS

The following deployment scenario definition has the main goal to achieve comparable results among different standards, standard proposals and vendor solutions. The scope of this parameter set is not to prefer a certain solution, but rather provide a common baseline parameter set. Although the parameter set defines a specific deployment scenario for evaluation purposes this does not imply that this scenario is necessarily in the focus of deployment by NGMN.

This deployment scenario tries to be as exact as possible without prescribing a specific technical solution, i.e. technology specific details (e.g. MIMO scheme) are not specified here, but should be exactly specified when the results are provided.

For each single parameter, good reasons may exist for the choice of a different value. For that reason, alternative values are given, which should be used only in rare occasions only to limit the parameter space.

Macro-cell baseline parameters for simulation purposes are shown in Table 3. They are closely aligned to [3]. The baseline value should at least be included in any simulations for calibration purposes.

Table 3 Macro-cell simulation baseline parameters

Parameter	Baseline Value	Alternative Value
User Traffic Model	Traffic Mix as described in Table 5	Full Buffer
Site Layout	3-sectorized Hexagonal grid with 19 cells and wrap-around	
Site-to-site distance	500m	1km, 1500m
Carrier frequency	2 GHz	2.5 GHz, 900 MHz
Operating Bandwidth	10 MHz per uplink or downlink for FDD deployment, 10 MHz for TDD deployment	2x20 MHz and 2x1.4 MHz for FDD deployment and 20 MHz for TDD deployment 2x5 MHz (FDD)or 1x10 MHz (TDD) for VoIP traffic evaluation
TDD Uplink/Downlink Ratio	As close as possible to a 18:29 UL:DL ratio for data and control symbols	
Distance-dependent pathloss	$L = I + 37.6\log_{10}(R)$, R in kilometers, $I = 128.1$ for 2 GHz	
Lognormal shadowing	"Gudmundson Model" as used in UMTS 30.03, B 1.4.1.4 (TR 101.112) [12]. Decorrelation length 50 meters	

Parameter	Baseline Value	Alternative Value
Shadowing Standard Deviation	8 dB	
Correlation distance of shadowing	50 m	
Shadowing correlation between cells	0.5	
Shadowing correlation between sectors	1.0	
Penetration loss	20 dB	
Horizontal antenna pattern for 3-sector cell sites with fixed antenna patterns	$A(\theta) = -\min\left[12\left(\frac{\theta}{\theta_{3dB}}\right)^2, A_m\right],$ $\theta_{3dB} = 70^\circ, A_m = 20dB$	
User location	Uniformly dropped in entire cell	
UE speeds of interest	3 km/h	30 km/h, 120 km/h, 350 km/h Or set of mixed speeds with 60% of users at 3km/h, 30% at 30km/h, and 10% at 120km/h)
Total BS Tx power	46 dBm (40W) for 10 MHz bandwidth	
Number of BS antennas per sector	2 for Rx and Tx (5 dB noise figure, 14 dBi BS antenna gain plus cable loss)	1[1],4,8 BS Rx and Tx antennas
BS antenna configuration	X-polarized (+45°/-45°) antenna with two ports	Cross-polarized linear array[2]. For 4x2 MIMO X-polarized and $\lambda/2$ spacing or vertically polarized antennas with $\lambda/2$ spacing
Minimum distance UE to BS	35m	
UE Tx power	24 dBm (250 mW)	
UE antennas	2 Rx antennas, 1 Tx antenna, 0 dBi antenna gain, 9 dB noise figure	2 Tx antennas, 4 Rx antennas
UE antenna configuration	Cross-polarized	

[1] For comparison reference only
[2] Antenna configuration should be described exactly, e.g. number of antenna columns, polarization, separation between antenna columns, number of antenna ports,

Parameter	Baseline Value	Alternative Value
Channel models without MIMO consideration	GSM Typical Urban (TU) [14]	
Channel models with MIMO consideration (SDMA, spatial multiplexing)	Urban Macro High Spread 3GPP/3GPP2 Spatial Channel Model (SCM) [13] and its extension to wider bandwidths	For mixed speed cases ITU-R channel models with spatial correlation

Additional parameters not included in the table should be quoted. These include the antenna heights at BS, UE, antenna pattern, BS antenna downtilt, noise figure, antenna gain, RF radio transceiver characteristics and the number of fixed and mobile relay stations (if used).

$$L = I + 37.6\log_{10}(R)$$

Specific technology configuration (like scheduler , frequency reuse scheme, link adaptation, modulation and coding, spatial multiplexing method, interference coordination, power control, maximum number of HARQ retransmissions etc.) should be described exactly. An example configuration can be found in [4].

Table 4 shows simulation assumptions.

Parameter	Value	Alternative Value
Channel estimation	Non-ideal	Ideal. The performance values should be corrected by an appropriate correction factor, if ideal channel estimation is used. These correction factors are usually different for uplink and downlink, and for different technologies.
Feedback channel errors	Non-ideal[3]	Ideal
Control channel errors	Non-ideal	Ideal
Control and Pilot Channel Overhead, Acknowledgements etc.	Exact[4]	

[3] The assumed feedback channel delay should be given
[4] The breakdown of overhead elements should be given

3 ANNEX

A TRAFFIC MODELS

The traffic mix given in the table below is in percentage of users. This criterion is used since it is favourable to simulate with state-of-the-art simulators.

Table 5 User Traffic Mix

Application	*Traffic Category*	*Percentage of Users*
FTP	Best effort	10 %
Web Browsing / HTTP	Interactive	20 %
Video Streaming	Streaming	20 %
VoIP	Real-time	30 %
Gaming	Interactive real-time	20 %

The traffic models outlined below (e.g. best effort – FTP, HTTP QoS – VoIP, streaming, video conferencing, gaming) should be considered. They are used by most standardization bodies, e.g. in and in [3] and in [4].

Satisfied user criteria for different the traffic types can be found in [3] in section 8.1.1.

Best Effort Traffic: FTP

The following definition is for the downlink. For the uplink, the same traffic model shall be used. An FTP session is a sequence of file transfers separated by reading times. The two main FTP session parameters are

The size S of a file to be transferred

The reading time D, i.e. the time interval between end of download of previous file and the user request for the next file

Table 6 FTP Traffic Parameters

Parameter	Statistical Characterization
File Size *S*	Truncated Lognormal Distribution Mean= 2Mbytes, Standard Deviation=0.722 Mbytes, Maximum=5 Mbytes (Before Truncation) PDF: $f_x = \frac{1}{\sqrt{2\pi}\sigma x} e^{\frac{-(\ln x - \mu)^2}{2\sigma^2}}, x > 0$, $\sigma = 0.35, \mu = 14.45$
Reading Time D	Exponential Distribution Mean=180 seconds PDF: $f_x = \lambda e^{-\lambda x}, x \geq 0$, $\lambda = 0.006$

Interactive Traffic: Web-browsing using HTTP

A web-page consists of a main object and embedded objects (e.g. pictures, advertisements etc). After receiving the main page, the web-browser will parse for the embedded objects. The main parameters to characterize web-browsing are:

- The main size of an object S_M
- The size of an embedded object in a page S_E
- The number of embedded objects N_D
- Reading time *D*
- Parsing Time for the min page T_P

Table 7 Web-browsing Traffic Parameters

Parameter	Statistical Characterization
Main Object Size S_M	Truncated Lognormal Distribution Mean=10710 Bytes, Standard Deviation=25032 Bytes, Minimum=100 Bytes, Maximum=2 Mbytes (Before Truncation) PDF: $f_x = \frac{1}{\sqrt{2\pi}\sigma x} e^{\frac{-(\ln x-\mu)^2}{2\sigma^2}}, x > 0$. $\sigma = 1.37, \mu = 8.37$
Embedded Object Size S_E	Truncated Lognormal Distribution Mean=7758 Bytes, Standard Deviation=126168 Bytes, Minimum=50 Bytes, Maximum=2 Mbytes (Before Truncation) PDF: $f_x = \frac{1}{\sqrt{2\pi}\sigma x} e^{\frac{-(\ln x-\mu)^2}{2\sigma^2}}, x > 0$. $\sigma = 2.36, \mu = 6.17$
Number of Embedded Objects per Page $=N_D$	Truncated Pareto Distribution Mean=5.64, Maximum=53 (Before Truncation) PDF: $f_x = \frac{\alpha_k^\alpha}{\alpha+1}, k \le x < m$. $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$. $\alpha = 1.1, k = 2, m = 55$ Note: Subtract k from the generated random value to obtain N_D
Reading Time D	Exponential Distribution Mean=30 seconds PDF: $f_x = \lambda e^{-\lambda x}, x \ge 0$. $\lambda = 0.033$
Parsing Time T_P	Exponential Distribution Mean=0.13 seconds PDF: $f_x = \lambda e^{-\lambda x}, x \ge 0$. $\lambda = 7.69$

Video Streaming

Each frame of video data arrives at a regular interval T determined by the number frames per second. Each frame is decomposed into a fixed number of slices, each transmitted as a single packet. The size of these packets/slices is modelled to have a truncated Pareto distribution. The video encoder introduces encoding delay intervals between the packets of a frame. These intervals are modelled by a truncated Pareto distribution. The following distributions assume a source video rate of 64 kbps:

Table 8 Video Streaming Traffic Parameters

Parameter	Statistical Characterization
Inter-Arrival time between the beginning of each frame	Deterministic 100 ms (based on 10 frames per second)
Number of packets (slices) in a frame	Deterministic, 8 packets per frame
Packet (slice) size	Truncated Pareto Distribution Mean=100 Bytes, Maximum =250 Bytes (Before Truncation) PDF: $f_x = \frac{\alpha_k^\alpha}{\alpha+1}, k \le x < m$, $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$, $\alpha = 1.2, k = 20 Bytes, m = ??$
Inter-arrival time between packets (slices) in a frame	Truncated Pareto Distribution Mean=*m*=6 ms, Maximum =12.5 ms (Before Truncation) PDF: $f_x = \frac{\alpha_k^\alpha}{\alpha+1}, k \le x < m$, $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$, $\alpha = 1.2, k = 2.5 ms, m = ??$

Interactive Real-Time Services: Gaming

Table 9 Uplink Gaming Network Traffic Parameters

Parameter	Statistical Characterization
Initial packet arrival	Uniform Distribution $f_x = \frac{1}{b-a}, a \le x \le b$, $a = 0, b = 40ms$
Packet arrival	Deterministic, 40 ms
Packet size	Largest Extreme Value Distribution (also known as Fisher-Tippett distribution) $f_x = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}$, $a = 45 Bytes, b = 5.7$ Values for this distribution can be generated by the following procedure: $x = a - b\ln(-\ln y)$, where y is drawn from a uniform distribution in the range $[0,1]$ Because packet size has to be integer number of bytes, the largest integer less than or equal to x is used as the actual packet size
UDP header	Deterministic (2 Bytes). This is added to the packet size accounting for the UDP header after header compression.

To simulate the random timing relationship between client traffic packet arrival and uplink frame boundary, the starting time of a network gaming mobile is uniformly distributed within [0,40 ms].

A maximum delay of 160 ms is applied to all uplink packets, i.e. a packet is dropped by the mobile station if any part of the packet has not started physical layer transmission, including HARQ operation, 160 ms after entering the mobile station buffer. The packet delay of a dropped packet is counted as 180 ms.

A mobile network gaming user is in outage if the average packet delay is greater than 60 ms. The average delay is the average of the delays of all packets, including the delay of packets delivered and the delay of packets dropped.

Table 10 Downlink Gaming Network Traffic Parameters

Parameter	Statistical Characterization
Initial packet arrival	Uniform Distribution PDF: $f_x = \frac{1}{b-a}, a \le x \le b$. $a = 0, b = 40ms$
Packet arrival	Largest Extreme Value Distribution (also known as Fisher-Tippett distribution) PDF: $f_x = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}$, $a = 55ms, b = 6$ Values for this distribution can be generated by the following procedure: $x = a - b\ln(-\ln y)$, where y is drawn from a uniform distribution in the range $[0,1]$
Packet size	Largest Extreme Value Distribution (also known as Fisher-Tippett distribution) $a = 120 Bytes, b = 36$
UDP header	Deterministic (2 Bytes). This is added to the packet size accounting for the UDP header after header compression.

B VOIP SATISFIED USER CRITERION AND TRAFFIC MODEL

Satisfied user criterion

A VoIP user is in outage (not satisfied) if 98% radio interface tail latency of this user is greater than 50 ms. This assumes an end-to-end delay below 200 ms for mobile-to-mobile communications.
The system capacity is defined as the number of users in the cell when more than 98% [5]of the users are satisfied.
Erasure rate for consecutive full rate AMR voice frames is proposed to be [less than 0.05% to be defined by SA4] (to account for bundled voice packet losses). Bundling is considered as an enhancement technique and the outage criterion used should be detailed for such evaluations.

[5] If simulators have limitations to provide statistically significant values, 95% of satisfied users can be selected alternatively.

Main parameters of the traffic model

The following table provides the relevant parameters of the VoIP traffic that shall be assumed in the simulations. The main purpose of this traffic model is not to favour any codec but to specify a model to obtain results which are comparable. The details of the corresponding traffic model are described below:

Parameter	*Characterization*
Codec	RTP AMR 12.2, Source rate 12.2 kbps
Encoder frame length	20 ms
Voice activity factor (VAF)	50% (c=0.01, d=0.99)
SID payload	Modelled 15 bytes (5Bytes + header) SID packet every 160ms during silence
Protocol Overhead with compressed header	10 bit + padding (RTP-pre-header) 4Byte (RTP/UDP/IP) 2 Byte (RLC/security) 16 bits (CRC)
Total voice payload on air interface	40bytes (AMR 12.2)

Detailed description of the VoIP traffic model

Basic Model

Consider the simple 2-state voice activity model shown in Figure 4.[6]

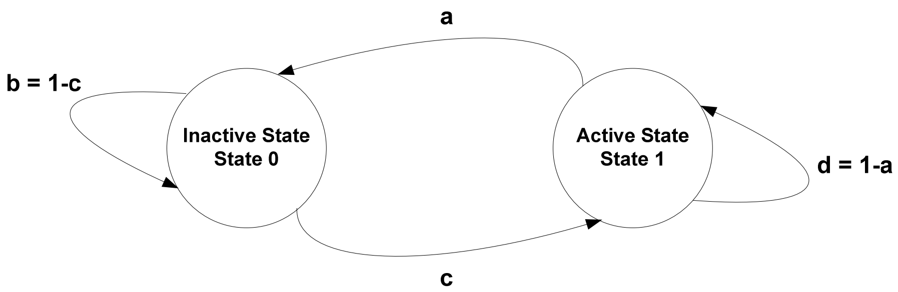

Figure 1 – 2-state voice activity model.

In the model, the probability of transitioning from state 1 (the active speech state) to state 0 (the inactive or silent state) while in state 1 is equal to a, while the probability of transitioning from state 0 to state 1 while in state 0 is c. The model is assumed updated at the speech encoder frame rate $R = 1/T$, where T is the encoder frame duration (typically, 20ms).

Model Statistics

The steady-state equilibrium of the model requires that

$$P_0 = \frac{a}{a+c} \qquad P_1 = \frac{c}{a+c} \tag{0.1}$$

where P_0 and P_1 are respectively the probability of being in state 0 and state 1.

The Voice Activity Factor (VAF) λ is given by

$$\lambda = P_1 = \frac{c}{a+c} \tag{0.2}$$

A talk-spurt is defined as the time period τ_{TS} between entering the active state (state 1) and leaving the active state. The probability that a talk spurt has duration n speech frames is given by

$$P_{\tau_{TS}=n} \triangleq P_{TS}(n) = a(1-a)^{n-1} \quad n = 1, 2, \cdots \tag{0.3}$$

Correspondingly, the probability that a silence period has duration n speech frames is given by

6 Clearly, a 2-state model is extremely simplistic, and many more complex models are available. However, it is amenable to rapid analysis and initial estimation of talk spurt arrival statistics and hence reservation activity.

$$P_{\tau_{SP}=n} \triangleq P_{SP}(n) = c(1-c)^{n-1} \quad n = 1, 2, \cdots \tag{0.4}$$

The mean talk spurt duration μ_{TS} (in speech frames) is given by

$$\mu_{TS} = E[\tau_{TS}] = \frac{1}{a} \tag{0.5}$$

while the mean silence period duration μ_{SP} (in speech frames) is given by

$$\mu_{SP} = E[\tau_{SP}] = \frac{1}{c} \tag{0.6}$$

The distribution of the time period τ_{AE} (in speech frames) between successive active state entries is the convolution of the distributions of τ_{SP} and τ_{TS}. This is given by

$$P_{\tau_{AE}=n} \triangleq P_{AE}(n) = \frac{c}{c-a} a(1-a)^{n-1} + \frac{a}{a-c} c(1-c)^{n-1} \quad n = 1, 2, \cdots \tag{0.7}$$

Note that τ_{AE} can be used as a crude guide to the time between MAC layer resource reservations, provided a single reservation is made per user per talk-spurt. Note that in practice, very small values of τ_{AE} might not lead to a separate reservation request, but equation (0.7) still offers some potentially useful guidance.

Since the state transitions from state 1 to state 0 and vice versa are independent, the mean time μ_{AE} between active state entries is given simply by the sum of the mean time in each state. That is

$$\mu_{AE} = \mu_{TS} + \mu_{SP} \tag{0.8}$$

Accordingly, the mean rate of arrival $\bar{R}_{AE}$ of transitions into the active state is given by

$$\bar{R}_{AE} = \frac{1}{\mu_{AE}} \tag{0.9}$$

Example

As a simple example, consider the case where the speech encoder frame duration $T = 20ms$. Further assume a desired VAF of 60% ($\lambda = 0.6$), and a desired mean talk spurt duration of 5s. Accordingly, from equation (0.5), $1/a = 5/T$ and so $a = 0.04$. Further, from equation (0.2), $c = a\lambda/(1-\lambda) = 0.006$.

For these parameters, the resulting theoretical and simulated distributions of the talk spurt duration (τ_{TS}, in seconds), silence period duration (τ_{SP}, in seconds), and time between active state entry (τ_{AE}, in seconds) appear in Figure 5.

The mean talk spurt duration is given by $\mu_{TS} = 1/a = 250$ frames, or 5s. Correspondingly, the mean silence period duration is $\mu_{SP} = 1/c = 166.67$ frames, or 3.33s. The resulting mean time between active state entry is then 8.33s, and so the mean rate of arrival of talk spurts is $\bar{R}_{AE} = 1/8.33 = 0.12$ talk spurts per second.

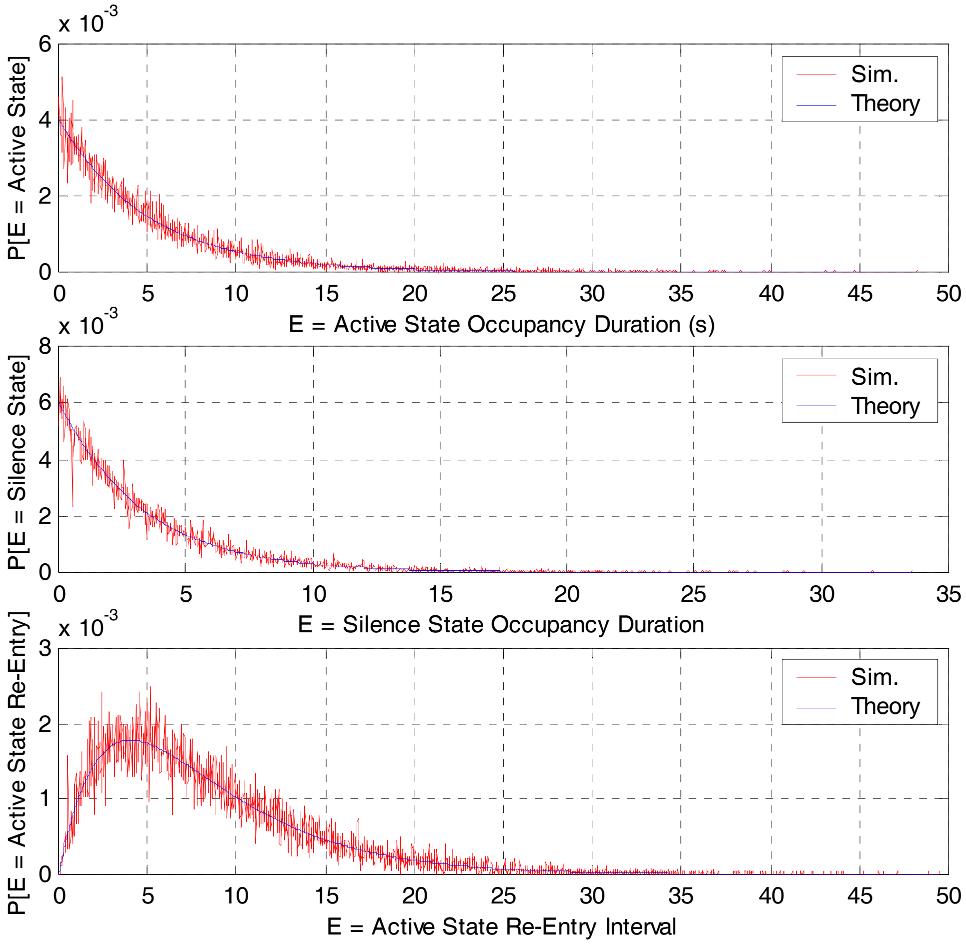

Figure 5 – State duration and entry distributions – theory and simulation.

C EXAMPLE CONFIGURATIONS

This chapter gives example configurations for standards addressing the NGMN requirements. The standards considered here are 3GPP LTE FDD, 3GPP LTE TDD, Mobile WiMAX Wave II Rel. 1.0, and UMB. This list of example standards does not imply that they are preferred by NGMN, or that this set is complete.

This evaluation methodology is meant to be technology-agnostic. Certain technology elements are defined for the purpose to establish a baseline assumption to allow for an apples-to-apples comparison of different simulation results. NGMN encourages the development of novel technology elements beyond these baseline values, if the requirements in [1] are addressed even better.

Table 12: Example simulation configuration

Parameter	*Baseline Value*	*Alternative value*
UE receiver structure	LMMSE	Interference Rejection Combining (IRC), Successive Interference Cancellation (SIC) or Maximum Likelihood Detector (MLD)
BS receiver structure	LMMSE	Interference Rejection Combining (IRC) or Maximum Ratio Combining (MRC)
Scheduler	Proportional Fair (PF) in time and frequency, if applicable	
Spatial Processing Schemes	Spatial Multiplexing	Adaptive switching between favourable schemes
Interference Treatment Method at BS and UE	None	Interference Rejection Combining (IRC) at BS and UE
Power Control	Slow uplink power control with 40 Transmit Time Interval (TTI) period	Open loop power control
HARQ Scheme	Chase combining, 8 TTI delay	Incremental redundancy (if supported by standard),
Network Synchronization	None	
Receiver noise floor	-174 dBm/Hz	
Link-to-system interface for simulations	Mutual Information Effective SNR Metric (MIESM) or Exponential Effective SNR Mapping (EESM)	
Frame length	5ms for WiMAX	
Link Adaptation	Included	
CQI Reporting	CQI feedback period of 5 ms with a delay of 2 ms	
Overhead		

Channel Estimation Correction Factors

Channel estimation correction factors were obtained empirically by a set of independent system simulations run by NGMN members and sponsors, which were averaged. The data rate / spectral efficiency obtained by a simulation which assumes ideal channel estimation has to be multiplied by the correction factor given below, to obtain a value comparable to results with realistic channel estimation.

Table 13: Channel Estimation Correction Factors

	DL		UL	
	Cell-average	**Cell-edge**	**Cell-average**	**Cell-edge**
WiMAX PUSC	*0.96*	*0.92*	*0.888*	*0.760*
WiMAX Band AMC	*0.94*	*0.93*	*0.938*	*0.836*
LTE TDD	*0.95*	*0.93*	*0.964*	*0.909*
LTE FDD	*0.97*	*0.95*	*0.938*	*0.836*

Overhead

Overhead due to control channels and reference symbols (pilots) is very significant in all technologies addressing NGMN requirements. Good modelling of overhead in simulations is very complex.

Overhead can be modelled in a static or dynamic way (e.g. varies during simulation depending on scheduling, traffic models, channel conditions etc.)

Overhead in the downlink contains cyclic prefix, guard subcarriers, pilots, control, synchronization broadcast channels, acknowledgements etc.

Overhead in the uplink contains additionally random access channels.

The following overhead breakdown should be seen as an example for different technologies. Standards are subject to change.

Overhead Estimation for WiMAX – Release 1 Wave 2

The following assumptions are made:

- 5 ms frame size, 10 MHz BW
- Over-sampling rate : 11.2 MHz
- Each frame comprises of 48 symbols
- Cyclic Prefix overhead: 1/8 of a symbol : 11.43 us
- Total number of data sub-carriers for DL PUSC : 720
- Number of overhead symbols: 1(TTG) +10(6DL MAP + 1DL Preamble+3UL) = 11 (with 6 symbols MAP overhead) or =9 with 4 symbols MAP overhead
- CP=1/8 x OFDM useful symbol duration is equivalent to utilizing 8/9th of an OFDMA symbol = 12.5%

We get then the following overhead components:

- DL Overhead Modelling (except pilots)
  - Preamble (1 symbol)
  - FCH
  - MAP (4-6 symbols), depending on
    - Fixed part +dynamic part (function of number of users scheduled/TDD frame)
    - Dynamic part depends upon number of user scheduled
    - Reuse factor
    - Traffic types
    - Transmit diversity using 2/4 Tx Antennas
    - Repetition factor (4 or 6)
  - DL Pilot Overhead (PUSC): 14.28%
- UL Overhead Modelling (excepts pilots)
  - Assuming symbols for Ranging + A/N + CQI
  - Indicate number of users that is multiplexed for Ranging + A/N + CQI
  - TTG+RTG : 1 symbol
  - Sounding signal : 1 symbol
- UL Pilot Overhead (PUSC): 33.3%
- Maximum UL+DL Overhead (except pilots)
  - (1+6+3+1) = 11 symbols if sounding not used : 22.9%
- CP Overhead 12.5%

Overhead Estimation for LTE FDD (as of 12/2007)

The following assumptions are made:

Overhead (DL):

- 1 ms frame size, 5 MHz BW
- Over-sampling rate: 7.68 MHz
- Each sub-frame comprises of 14 symbols and 300 resource elements (RE) per symbol
- Total Cyclic Prefix overhead/0.5 ms slot: 33 us which is 6.6% overhead
- First 2 or 3 symbols comprises of control+pilot = 600 or 900 RE's (worst case)
- 5th, 8th and 12th symbol comprises additional pilots for antenna 1 and 2 = 100x3 = 300 RE
- Total RE's for SCH+BCH ~ 58RE's
- Total number of data sub-carriers (RE) in DL: 300x14 – 900 – 100x3 - 58 = 2942
- Number of overhead RE's: 1258 RE's (worst case) or (958)

With that, it leads to the following overhead components:

- DL Overhead Modelling
  - Total of 300 RE's/symbol, 14 symbols/sub-frame(1ms)
  - DL Reference Symbol Overhead: 2 Tx :9.52%, 4Tx:14.29%
  - DL Control Channel overhead: Explicitly modeled
    - Number of symbols can be N=2 or 3
    - Worst case N=3 : 19%, N=2 : 11.9%
  - SCH Overhead: 288 RE's per 10 ms : 0.7%
  - BCH Overhead: 288 RE's per 10 ms : 0.7%

- UL Overhead Modelling
  - Total of 300 RE's/symbol, 14 symbols/sub-frame(1ms)
  - UL Reference Symbol Overhead: 14.3%
  - UL Control Channel: 8%
  - UL RACH: 1 RACH/10ms: 2.4%
  - UL SRS : 1 symbol/sub-frame/5 ms: 1.4%

- CP overhead 6.6%

Overhead Estimation for LTE TDD (as of 12/2007)

The following assumptions are made:

Overhead (DL):

- 1 ms sub-frame size, 0.5 ms slot size, 10 MHz BW
- Over-sampling Rate : 15.36 MHz
- Each sub-frame comprises of 14 symbols and 600 resource elements (RE) per symbol,
- Total Cyclic Prefix overhead/0.5 ms slot: 33 us which is 6.6% overhead
- First 2 or 3 symbols comprises of control+pilot = 1200 or 1800 RE's (worst case)
- 5th, 8th and 12th symbol comprises additional pilots for antenna 1 and 2 = 200x3 = 600 RE (2 antennas)
- Total RE's for S-SCH+BCH on TS0 ~ 72RE+600 REx4
- Total number of data sub-carriers (RE) in DL: 600x14 – 1200 – 200x3 = 6600, 6000(worst case)

Overhead (UL)

- 1 ms frame size, 10 MHz BW
- Over-sampling Rate : 15.36 MHz
- Each sub-frame comprises of 14 symbols and 600 resource elements (RE) per symbol,
- Total Cyclic Prefix overhead/0.5 ms slot: 33 us which is 6.6% overhead
- 2 edge RB are reserved for control (ACK/NACK, CQI etc) which is equivalent to 2x12x12 = 288 RE
- 2 LB pilots is equivalent to 1200 RE
- UL SRS : 1 symbol/sub-frame/5 ms
- Total number of data sub-carriers (RE) in UL: 600x14 – 1200 – 24x12 - 600

Overhead (Special Slot)

- 1 ms slot size, 10 MHz BW,
- Option1 : Short RACH covers 2 symbols, 6RB, 4 PRACH channel, total =4x72x2=576RE
- Option2: no short RACH when UpPTS covers 1 symbol,
- P-SCH covers 1 symbol, 6RB, total=71 RE
- Normal GP covers 2 symbols, total = 600x2=1200RE
  - or Normal GP covers 1 symbols
  - total= 600RE
- Pilots on special slot not decided.
- Total number of symbols of dwPTS, upPTS and GP is not yet defined.
- Total number of REs in dwPTS should exclude P-SCH overhead and pilots.
- Total number of REs in upPTS should exclude RACH overhead and pilots.

With that, it leads to the following overhead components:

- DL Overhead Modelling
  - Total of 600 RE's/symbol, 14 symbols/sub-frame(1ms)
  - DL Reference Symbol Overhead: 2 Tx :9.52%, 4Tx:14.29%
  - DL Control Channel overhead: Explicitly modeled
    - Number of symbols can be N=2 or 3
    - Worst case N=3 : 19%, N=2 : 11.9%
  - SCH Overhead: 288 RE's per 10 ms : 0.7%
  - BCH Overhead: 288 RE's per 10 ms : 0.7%
- UL Overhead Modelling
  - Total of 600 RE's/symbol, 14 symbols/sub-frame(1ms)
  - UL Reference Symbol Overhead: 14.3%
  - UL Control Channel: 8%
  - UL RACH: 1 RACH/10ms: 1.2%
  - UL short RACH: 1 short RACH/5ms: 2.8% , 10ms: 1.4%
- CP overhead 6.6%
- GP overhead

It should be noted that developments of these standards are ongoing and these values reflect the status of these standards by December 2007.

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E ABBREVIATIONS

3GPP	Third Generation Partnership Project
3GPP2	Third Generation Partnership Project 2
AMR	Adaptive Multi Rate
BS	Base Station
CDF	Complementary Distribution Function
C-Plane	Control Plane
E2E	End to End
FDD	Frequency Division Duplex
IEEE	Institute of Electrical and Electronics Engineers
ISD	Inter Site Distance
HARQ	Hybrid ARQ
L1/L2/L3	Layer 1/2/3
LTE	Long Term Evolution
MAC	Media Access Control
Mbps	Megabits per second
MBMS	Multimedia Broadcast Multicast Service
MIMO	Multiple Input Multiple Output
NGMN	Next Generation Mobile Networks

PER	Packet Error Rate
PHY	Physical Layer
QoS	Quality of Service
PDF	Probability Density Function
RAN	Radio Access Network
RE	Resource Element
RF	Radio Frequency
RLC	Radio Link Control
SDMA	Space Division Multiple Access
SFN	Single Frequency Network
TCP	Transmission Control Protocol
TDD	Time Division Duplex
Tx	Transmit
UE	User Equipment
U-Plane	User Plane
UTRAN	UMTS Terrestrial Radio Access Network
VoIP	Voice over IP