A QoS Concept for Packet Oriented S-UMTS Services

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ABSTRACT

An approach by the IST FUTURE project to make packet based multimedia services feasible in a satellite based UMTS environment. Different QoS guaranteeing mechanisms are presented exploiting the terrestrial UMTS specification in order to achieve an integrated T-S-UMTS. The focus is set on mechanisms for the access network, while the core network is only theoretically investigated in order to give an end-to-end QoS approach. For the access network, based on the UMTS specifications, algorithms for Connection Admission Control, Scheduling, Active Set Handling and Measurement Control have been proposed for an S-UMTS environment.

I. Introduction

The IST FUTURE project is the logical continuation of the IST VIRTUOUS project on the migration path towards an integrated satellite and terrestrial based UMTS (T-S-UMTS). Thereby FUTURE aims at the integration of multimedia services and at the development of enhanced Quality of Service (QoS) strategies for a packet based UMTS to fulfil the requirements of the integrated services. Therefore the objective is to demonstrate the achievement of end-to-end QoS for the following services: Voice over IP (VoIP), web browsing and file transfer via ftp.

For the satellite segment the general consensus of the industry is towards a GEO (Geostationary Earth Orbit) based solution for S-UMTS, which means that the system has to deal with high delays and Bit Error Rates (BERs).

The topic of this paper is the FUTURE QoS concept which has to deal with the satellite channel and to be in compliance with the UMTS specifications at the same time. Thus after the introduction the requirements will be described in II. followed by a description of the different QoS guaranteeing algorithms which have been developed for the satellite access network, namely Connection Admission Control (CAC), Scheduling and Active Set Handling which are all supported by a measurement control unit as described in III. For an end-to-end QoS approach a short investigation of the Core Network (CN) for an integrated T-S-UMTS environment is given in IV.

II. Requirements of QoS in FUTURE

The QoS concept of FUTURE is concerning the S-UMTS access network (USRAN – UMTS Satellite Radio Access Network). This means that the standardized terrestrial UMTS QoS concept is taken as a basis for the FUTURE QoS concept as well. Thus FUTURE has adopted the four QoS classes and the layered structure as proposed by 3GPP for UMTS [1]. Only the parts relevant to the Radio Access Network (RAN) have been investigated and will be implemented For the CN only theoretical investigation is foreseen.

A. *FUTURE in the UMTS QoS architecture*

The whole FUTURE concept is based on the UMTS QoS concept and architecture as proposed by 3GPP in [1]. In UMTS all network services are considered end-to-end as shown in Figure 1. This end-to-end service concept was divided into different sublayers called Bearer Services (BS). Thereby each layer offers its services to the higher layer. As depicted in Figure 1 the FUTURE QoS concept focuses on the UMTS BS for the satellite segment, which means that the URAN in Figure 1 will be a USRAN (UMTS Satellite Access Network). Consequently the BS covering the USRAN will be investigated and implemented by FUTURE.

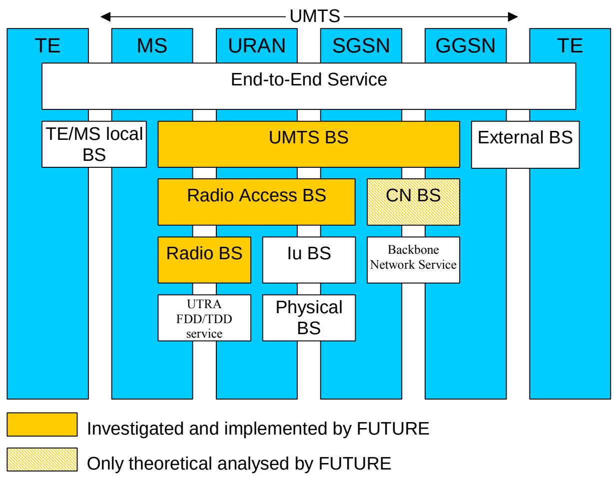

Figure 1: UMTS Bearer support in FUTURE

B. *The UMTS QoS classes*

In order to allow an easier development of QoS guaranteeing algorithms all services will be divided in four defined QoS classes:

- Conversational class
- Streaming class
- Interactive class
- Background class

These four classes can be mainly distinguished by their delay sensitiveness and coincide with the traffic classes which have been introduced in the GPRS specifications. Concerning the QoS constraints the specific characteristics of the satellite segment have to be taken into account, i.e., it has also to be considered that the satellite segment will be in all likelihood GEO based, which means with extreme long delays and high BER. In Table 1 some ranges for these parameters in the different environments are given as proposed by the 3GPP in [2]. In the first row some reference values for the satellite segment are give whilst all the others are concerning the terrestrial segment.

	Real Time		Non Real Time	
	BER	**Max. Delay**	**BER**	**Max. Delay**
Satellite	10^{-3}-10^{-7}	< 400 ms	10^{-5}-10^{-8}	> 1200ms
Rural	10^{-3}-10^{-7}	20-300 ms	10^{-5}-10^{-8}	> 150 ms
Urban	10^{-3}-10^{-7}	20-300 ms	10^{-5}-10^{-8}	> 150 ms
Indoor	10^{-3}-10^{-7}	20-300 ms	10^{-5}-10^{-8}	> 150 ms

Table 1: Range of QoS requirements

App.	Data Rate	Delay	Delay Variation	Information Loss
VoIP	4-25 kbps	<150ms (preferred) <400ms (limit)	< 1ms	< 3% FER
WWW		<4s / page	N.A.	Zero
FTP		< 10s	N.A.	Zero
Streaming	32-384 kbps	< 10s	< 1ms	< 1% FER

Table 2: End-user Performance Expectations

In Table 2 some values are proposed for different applications as they are going to be demonstrated within the FUTURE project. In both tables it can be seen that the delay constraints for real time services, especially in the conversational class, which means Voice over IP (VoIP) are very critical. Considering that the propagation delay for a GEO constellation is about 240ms without computing and waiting time this constraints are hard to fulfil. Thus good QoS guaranteeing algorithms for an efficient resource utilisation are needed. In the other classes represented by Web-Browsing (WWW), file transfer (FTP) and streaming the values are not so critical, which results in a better flexibility for the QoS algorithms. The streaming application mentioned in Table 2 includes both video and audio streaming whereby for streaming audio a maximum data rate of 128 kbps is foreseen.

III. QoS guaranteeing algorithms for S-UMTS Access Network

The QoS guaranteeing functions investigated in this section all concern the lower layers of the USRAN. In Figure 2 the QoS units of FUTURE are shown embedded in the UMTS architecture. It can be seen that CAC unit is placed on top of the architecture as it is responsible for allowing or refusing incoming calls based on parameters that are stored by the Measurement Control unit. It has to be mentioned that the congestion control elements are not part of FUTURE, but they are depicted for the sake of completeness as they have been part of the predecessor project VIRTUOUS.

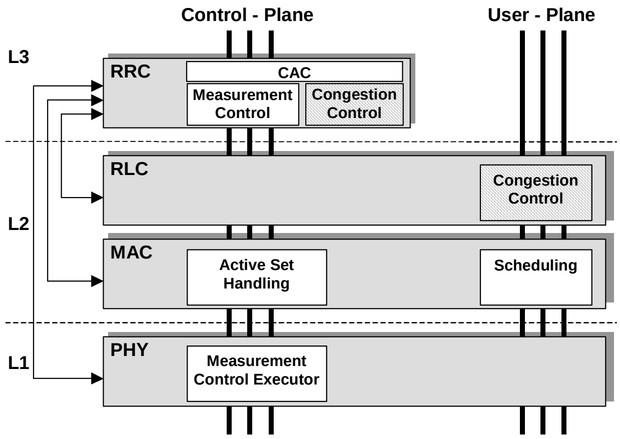

Figure 2: FUTURE QoS units within URAN

A. *Connection Admission Control*

Apparently, no admission request could be satisfied if the electromagnetic coverage conditions, at the point the requesting MN resides, are not adequate.

In FUTURE three connection admission control criteria will be implemented namely the Wideband Power-Based Admission Control Strategy, the Throughput-Based Admission Control Strategy [3][4], and CAC based on the signal-to-noise-plus-interference ratio either of which can be adopted in order for the CAC process to be realized at the Radio Bearer layer.

Wideband Power-Based Admission Control Strategy

According to this algorithm the criterion for the uplink admission of the connection is based on the interference the new user would add to the system, if admitted. For this reason an interference threshold value $I_{threshold}$ is set that should not be overpassed because of the estimated interference expected by the admission of the new connection request.

$$I_{total\_old} + \Delta I > I_{threshold}$$

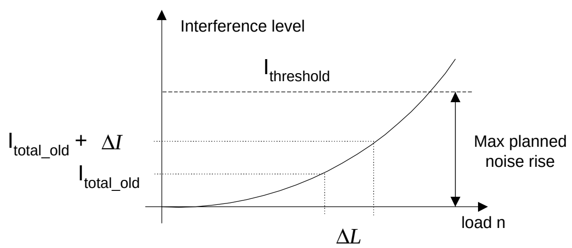

Figure 3: Load curve and the, due to a new connection, interference increase.

The value of the estimated increase of interfernce, ΔI, can be calculated by means of derivation or integration of the of the load curve, Figure 3.
In the first case (derivation) the estimated increase of interference is:

$$\Delta I \approx \frac{I_{total}}{1-n_{UL}}\Delta L$$

Where I_{total} is the total estimated interference level after admission of the new user, n_{UL} is the uplink load factor in the cell serving N users, ΔL is the load factor increase due to the new user. Therby n_{UL} is given by the following relation:

$$n_{UL} = (1+i)\cdot\sum_{j=1}^{N}\frac{1}{1+\frac{W}{(E_b/N_0)\cdot R_j\cdot v_j}}$$

With

$$i = \frac{\text{other cell interference}}{\text{own cell interference}}$$

W is the chip rate, Rj is the bitrate of the *j*th user and ΔL is given by the relation:

$$\Delta L = \frac{1}{1+\frac{W}{v\cdot E_b/N_0\cdot R}}$$

Where the bitrate, R, of the new connection dependents on service asked. The Signal energy per bit divided by noise spectral density (E_b/N_o) has to meet a predefined QoS (e.g. bit error rate). The noise includes both thermal noise and interference. The activity factor of the user v can be considered 0.67 for speech and 1.0 for data.
In the second case (integration) the estimated increase of interference is

$$\Delta I = \frac{I_{total}}{1-nul-\Delta L}\Delta L$$

All the parameters involved in this relationship have already been introduced and defined above.
Note, however, that the load is a function of the bandwidth asked.
Considering the downlink direction, the user is admitted if the new total downlink transmission power does not exceed a predefined target value set by the network operator:

$$P_{total\_old} + \Delta P_{total} > P_{threshold}$$

The load increase in the downlink can be estimated on the base of the initial power, which depends on the distance form the base station and is determined by the open loop control algorithm.

Throughput - Based admission Control Strategy

In throughput-based admission control strategy the new user is not admitted into the radio access network if

$$n_{UL} + \Delta L > n_{UL\_threshold}$$

and the same for downlink:

$$n_{DL} + \Delta L > n_{DL\_threshold}$$

Where the increase of load factor, ΔL, is calculated as in the previous algorithm. Moreover, n_{UL} and n_{DL} are the uplink and downlink factors before the admittance of the new connection. The n_{UL} is calculated as in the previous algorithm, while the n_{DL} is calculated as follows:

$$n_{DL} = \sum_{j=1}^{N} R_j\cdot\frac{v_j(E_b/E_0)_j}{W}\cdot[(1-\alpha_{av})+i_{av}]$$

α_{av}: the average orthogonality of the cell

i_{av}: the average downlink other–to-own cell interference ratio of the cell

CAC based on the signal-to-noise-plus-interference ratio

This admission control algorithm aims at preserving the quality of the connections measured in terms of the signal-to-noise-plus-interference ratio. We distinguish the operation on the uplink and downlink directions.

Uplink algorithm

The total power that a given base station (BS) antenna receives is compiled by the background noise, denoted by *N*, and the signals from MTs connected to the considered or nearby cells, denoted by *I* (interference). Let C_i and R_i be respectively the power and bit rate of the signal received from the *i*th MT connected to the base station. We will assume just one active connection per MT at the same time, although results can be easily extended to consider several connections with different QoS requirements. The bit energy to noise plus interference density ratio is given by:

$$\frac{E_b}{N_0+I_0} = \frac{C_i/R_i}{(N+I-C_i)/W} = \frac{C_i PG_i}{(N+I_i)}, \qquad (1)$$

where *W* is the chip rate of the system, PG_i is the so called *processing gain* and $I_i = I - C_i$ is the interference experimented by the user *i*. Hereafter, the interference density will be considered to be included in the noise density.
Upon a new MT connection request with a specific QoS demand, URAN will estimate the power to be received from the user to comply with the QoS error constraint, usually given in terms of BER or FER. A previous step is to derive a target E_b/N_0 that guarantees the error ratio required by the user. Among others, factors such as modulation parameters, error correction techniques, geographical location and MT movement pattern are used to map BER or FER specifications into the target E_b/N_0.
Assuming *M–1* users currently connected to the BS, the requesting user is the potential *M*th connected user. From (1), the minimum estimated required power for the new user is:

$$\tilde{C}_M = \frac{(E_b/N_0)_{target,M}(I_M+N)}{PG_M} \qquad (2)$$

I_M is the interference the new user would see if accepted, so $I_M + N$ is the whole power that the BS is receiving, i.e. the RSSI physical measurement in the UMTS standard [6]. Let $\tilde{C}_i$, $1 \le i \le M-1$, be the minimum new received power predicted for the *i*th MT which is required to meet its target E_b/N_0 with the effect of the requesting terminal taken into account. Under the assumption that the requested terminal is in the system, $\widetilde{C}_i > C_i$, where C_i is the current received signal power from the i th terminal. As the number of newly admitted mobile terminal increases, so should do the received power of the previously admitted ones, so that their required E_b/N_0 can be guaranteed. In turn, if $\widetilde{C}_i$, $1 \le i \le M-1$, increases, the total interference that the requesting terminal will experience also increases; hence, $\widetilde{C}_M$ needs to be increased. The rise of $\widetilde{C}_M$ will further increase the required $\widetilde{C}_i$ value. As a result, the predicted values need to be updated recursively based on the current interference seen by each user. After a few iterations, the estimations will converge with a reasonable accuracy if solution indeed exists. If the solutions diverge after a few iterations, it means that the system does not have enough capacity to accommodate the requesting MT. Another reason for failing is reaching the maximum transmitted power for a mobile terminal, being necessary the estimation of the path gain to check such condition.

Finally, if an admissible solution is found, the cell load factor might be calculated with the value of interference obtained. The new cell load factor must be under the maximum allowable load factor set in network planning. A high load factor could cause dropping calls if minor changes in the network state occur.

Downlink algorithm

As stated in the introduction, the downlink is limited by power availability rather than by interference. In S-UMTS, the downlink becomes the capacity bottleneck of the system.

In the same way it was done for the uplink, error requirements are mapped into a target E_b/N_0 to be achieved at the MT, which we will relate this time to the power to be transmitted by the base station. Starting from (1) a new equation for the received power at BS from user i can be derived:

$$C_i = \frac{I+N}{1+(PG_i/(E_b/N_0)_i)} \tag{3}$$

We will use an equivalent equation for the downlink:

$$C_i = \frac{(I+N)_i}{1+(PG_i/(E_b/N_0)_i)} \tag{4}$$

Now, C_i is the power with which the ith user channel is received at ith MT, and $(I+N)_i$ is the sum of total interference (C_i included) and background noise received at ith MT. Thus, when the Mth connection is requested, the estimation of its needed received power is:

$$\widetilde{C}_M = \frac{(I+N)_M}{1+(PG_M/(E_b/N_0)_{target,M})} \tag{5}$$

An alternative expression for $\widetilde{C}_M$ can be written in terms of the available power at the base station for user dedicated transmissions, P_{budget}, and the path loss from the BS to the MT, which can be obtained from measurements on an already active channel, e.g. the pilot channel:

$$\widetilde{C}_M = \frac{P_{budget}\widetilde{\phi}_M P_{RX\,pilot,M}}{P_{TX\,pilot}}, \tag{6}$$

$\widetilde{\phi}_M$ is the fraction of P_{budget} to be given to the Mth user; while $P_{TXpilot}$ and $P_{RXpilot,M}$ are the transmitted pilot power and the captured pilot power at Mth MT. Combining (5) and (6), the power fraction estimated to meet the user requirements is:

$$\widetilde{\phi}_M = \frac{RSSI_M P_{TX\,pilot}}{1+(PG_{Mi}/(Eb/N_0)_{target,M})P_{budget}P_{RX\,pilot,M}}$$

Under the assumption that the estimated power is admissible, the same problem we had in uplink arises now. If the connection is accepted and included in the system, the power transmitted by the BS increases, and so does the interference seen by the $M\!-\!1$ previously admitted connections, what can cause not respecting their QoS constraints. In the same way we did in the uplink, recursive calculations need to be done in order to estimate the situation in which the M users meet their requirements. If an admissible solution is found without reaching the maximum budgeted transmission power, the connection is admitted in the downlink.

The above described algorithm actually focus on the quality of the connection measured in terms of energy-to-noise ratio, aiming at guaranteeing a given error rate. In regard to the delay, bit rate and jitter constraints, the CAC will need the collaboration of the scheduler algorithm which will provide the information of whether it is able to respect such parameters in the current network state.

In the three algorithms described above, the interference from nearby cells was considered in the target cell, but the effect of the possible admission in nearby cells was not. To be admitted, the interference generated in every neighbouring cell must not exceed its current resource margin. Therefore, the target cell should transmit connection requirements to its neighbouring cells and receive permission from them.

Another factor to take into account is the fact that a connection request can be a new connection but it can also come from a handoff. The latter case should be prioritised because, in general , interrupting a service in an active connection is more annoying to users than rejecting a new connection. A fraction of the total resources can be reserved in a target cell for future handoff connections coming from nearby cells. The optimum fraction should depend on the handoff probability, which can be estimated from the traffic load in nearby cells, as well as measurements on the position, movement direction and mobility pattern of neighbouring MTs [7].

In satellite systems, handoffs are expected to be not so frequent as in terrestrial systems, but another inter-cell factor becomes essential: If it is possible to share the available power of the satellite between beams covering different cells, suitable algorithms should be designed to optimise power utilization for the whole population of users covered by a satellite.

All the above mentioned inter-cell considerations will not be implemented in the demonstrator due to existence

of only two users, but will be taken into account for the final UMTS architecture that the project will yield.
The FUTURE project will carry out the implementation of the three CAC algorithms and provide a critical assessment on the strengths an weakness of each one. To this end, performance metrics such as dropping probability, average accepted load and time taken in the in admission decision will be considered.

B. Scheduling

The main purpose of the Scheduling management function is to optimise the radio resources utilisation, and in particular maximising the amount of accepted requests and the data flow throughput, as well as satisfying given QoS parameters. In order to respect the Radio Access QoS sub-contract, the MAC scheduling function assigns W-CDMA codes and radio frames to data packets so that contracted QoS specifications (involving such indicators as delay, jitter, packet loss) are met. Among the possible CDMA techniques, FUTURE adoptes an OVSF scheme in which codes are organized into a code tree, and each user is assigned a single orthogonal code in the tree. We consider a scenario in which different services and/or applications can simultaneously be provided to two mobile users by a unique physical channel (Downlink Shared Channel, DSCH) per satellite beam.
In this context, the scheduling function has to solve the following problem:

Assign a suitable TF (Transport Format) to the channel and an OVSF code to each user so as to respect the QoS requirements.

In what follows, we present the main steps of an algorithm that can be used to implement the scheduling function.
For each user, a specific priority queue of packets is defined for each service typology (VoIP, VoD, WEB, FTP); the priority associated with a packet in a queue represents the relative importance of the packet, and/or the tightness of the relevant delay/jitter constraints.
A radio frame consists of a certain number of Packet Data Units (PDUs) and is trasmitted every 10ms (Minimum Interleaving Period, MIP). In general, a radio frame contains PDUs made of packets belonging to distinct users and having different priorities. Since in each satellite beam the two mobile users share the same radio resource, we divide each frame *f* into two half-frames *lf* and *rf* so that exactly one of the following holds:

- both *lf* and *rf* are assigned to the same user, and in this case for the MIP duration the channel is completely dedicated to that user;
- *lf* is assigned to one user and *rf* to the other, and in this case for the MIP duration each user is assigned half channel.

(Notice that unlike a frame, a half-frame is never shared between the users).
The first step of the algorithm consists in computing the best TF for the channel: this is a format allowing to send the maximum amount of information in the considered time horizon. The TF identifies the MAC PDU size (expressed in bits) and, consequently, the number of PDUs that are contained in a half-frame and will be transmitted by the shared transport channel.
Steps two and three are separately executed for the two users.
The second step consists in concatenating the packets of each PDU. Notice that the concatenation is feasible only between packets belonging to the same queue and with the same priority (the latter requirement allows to assign a single, definite priority to each PDU). To concatenate packets, we solve a Bin Packing problem involving all the packets of a given service typology and with a given priority. For each pair (typology, priority) we define:

$$x_{ij} = \begin{cases} 1 \text{ if packet } i \text{ is assigned to PDU } j \\ 0 \text{ otherwise} \end{cases}$$

$$y_j = \begin{cases} 1 \text{ if } \exists \text{ packet assigned to PDU } j \\ 0 \text{ otherwise} \end{cases}$$

$m =$# PDUs
$n =$# Packets
c_i = size of packet i.

An Integer Linear Programming (ILP) formulation of the problem is:

$$\min \sum_{j=1}^{m} y_j$$

$$\sum_{j=1}^{m} x_{ij} = 1 \quad \forall i = 1......n;$$

$$\sum_{i=1}^{n} c_i x_{ij} \le b_j y_j \quad \forall j = 1......m;$$

$$y_j \in \{0,1\}$$

$$x_{ij} \in \{0,1\}$$

Fast and effective heuristics for this problem are available in the literature, see e.g. [8].
In the third step, PDUs with the same priority are chosen to form half-frames (this operation is straightforward).
The fourth step takes as an input the user half-frames previously formed, and assigns them to the radio channel. This problem can be formulated as a *competitive matching problem* as follows. Define a bipartite graph *G* by placing a node a_i for each half-frame of user *A*, a node b_i for each half-frame of user *B* and two nodes u_t, v_t denoting the two halves of the channel available in the *t*-th MIP. Then place an edge between a_i (b_i) and u_t, v_t whenever the priority and characteristics of the *i*-th half frame of user *A* (*B*) allow it to be transmitted in the *t*-th MIP. A feasible assignment of the radio resource corresponds then to a matching of graph *G*.
To distinguish among solutions of different quality, we define for each edge *xy* of *G* a weight w_{xy} based on the

priority of the corresponding half-frame. The classical criterion of minimising the weight of a bipartite matching is met by several algorithms (the most suitable in this context appear to be on-line, see [9]). However, this criterion may not be sensible in our case due to the presence of users with possibly contrasting objectives: in fact, a minimum matching might penalize too much one of the two users. Thus, we resort to a different approach, based on the definition of *competitive matching*. A competitive matching is a matching such that the weight of the edges associated with user *A* is minimised, provided that the weight of the remaining edges of the matching (that is, those corresponding to user *B*) is not greater than a given threshold *k*. In many cases of interest this problem can be efficiently solved by dynamic programming.

C. Active Set Handling

The Active Set Handling for the satellite segment (S-UMTS), is based on the satellite diversity concept.

Satellite diversity can provide benefits in terms of reduced blockage probability, soft and softer-handoff capability, slow fading counteraction, and under certain conditions even increased system capacity. The probability of complete blockage greatly reduces with the number of satellites in simultaneous view, translating immediately into improved quality of service. The Active Set Handling procedure for QoS provision is clearly depending on the considered satellite constellation (LEO,GEO systems). This does not happen only because of the impact on propagation delay of different constellation but also because of the different handoff rate which may results for a LEO rather than GEO system. As a matter of fact, in the latter we may assume that the handoff rate is negligible for all users, apart for the aeronautical ones.

The case of double satellite diversity has been considered with the assumption that a clear path for both satellites was available.

In CDMA S-UMTS context, the soft satellite-handoff is realized through the Active Set Handling by updating the original monitored beams set, when needed. The satellite diversity mentioned above still retains the principles of FDD CDMA (or WCDMA), and it acts differently in forward and return link. In the forward link a satellite operator forces the system to practice diversity scheme just sending the same signal onto different satellites, with highly directive antennas. In the reverse link we do not have macro-diversity through satellite because of the UE's quasi-omni directional antennas.

Unlike T-UMTS, S-UMTS presents non-selective satellite fading channel, which preserves multiplex orthogonality and minimizes intra-beam interference. On the other hand, likewise the T-UMTS case, the signal replies sent by the Gateway to different non co-located satellite, introduce inter-beam interference, producing, in its turn, a capacity loss. This loss could be taken into account and kept with acceptable boundaries.

The Active Set Handling allows diversity in the forward link, permitting exploitation of antenna arrays and rake receiver in the UE-side. It operates in S-UMTS as a countermeasure to blockage-induced outage due to the on/off propagation channel properties. The concept of the active set is strictly related to the soft-handover one: soft-handover is a handover in which the UE starts communication with new satellite beam, on the same carrier, frequency, or sector of the same site (softer handover), performing utmost a change of code. Based on the measurement of the set of cells monitored, the soft-handover function evaluates if any satellite beam should be add to (radio link addition), removes from (radio link removal), or replaces in (combined radio link addition and removal) the active set, performing Active Set Update.

The difference between the proposed S-UMTS active set handling algorithm and the T-UMTS one lies in the time scale considered for incoming measures and/or in threshold introduced for different monitored beams. On the other hand, the actions taken into account by these algorithms are the same.

The radio-link addition event is reported if the measure quality of a satellite beam is continuously above add_threshold (fixed in the algorithm) for a period of ΔT and the current active set is not full (maximum number of satellite beams allowed for the radio link). The radio link removal event is reported if the measure quality of a cell is continuously worst than remove_threshold for a period of ΔT. The combined radio link addition and removal event is reported if the measure quality of a candidate (outside the active set) satellite beam is continuously better than the quality of a beam monitored inside the active set for a period of ΔT.

The size adopted for the Active Set may vary, but usually it ranges from 1 to 3 beams. The upper value is conceived in order to avoid not only RF interferences and CDMA code depletion, but also to exploit low power level consumptions.

D. Measurement Control

In order to support the provision of QoS, the status of the system resources must be suitably monitored and tracked and a set of parameters/measurements relevant for the different QoS functionalities must be provided. Our attention is restricted to the measurements required to support the management of QoS for the S-UMTS case on the air interface (Uu interface).

A peculiarity of S-UMTS with respect to T-UMTS is that typically in the satellite case the down-link is power-limited instead of bandwidth (interference) limited as in the terrestrial case. Forward Link capacity is thus directly related to available on-board RF power and Admission Control has thus to be based (although not exclusively) on the estimation of available RF power and evaluation of the adequacy of such power for supporting the requested QoS.

To this end of fundamental importance are measurements of the received power on pilot channels at the MT. Based on the guaranteed bit rate, the target quality (BER and SDU error rate) and the MT report about the received power on pilot channels the

Admission Control procedure performs an estimation of the required RF power to support the service.
As far as scheduling is concerned, due to the random nature of the propagation channel, the outage condition and/or the bad propagation state has to be considered. For the satellite case the propagation channel may likely be characterized as on-off due to fact that satellite communication are restricted to operate only in presence of a line-of-sight path. A *Link Status* condition shall be thus derived from the received power measurements performed at the MT.
In the reverse link there is virtually no limitation coming from on-board power and from code limitation. Interference and MT EIRP are the real capacity bottleneck.
The most appropriate system loading parameter is represented in such a case by the *Beam Received Wide-band Power* consisting of the wide-band power received on each frequency slot of relevant beam. Knowing the MT characteristics (in particular the EIRP), it is possible to predict if sufficient power is available at the MT to support the requested service and if the increase of interference caused by the new allocation can be sustained by the other users (not only of the beam /satellite but also on adjacent beams or other visible satellites).

Physical Layer Measurements

Based on these measurements further higher-level system parameters may be derived. In particular the following measurements will be performed at physical layer level.

RSSI (Received Signal Strength Indicator)
This measurement is performed by the MT demodulators and consists of the total power (signal, thermal noise and interference) received. The most convenient reference point for the measurement is the output of the Chip Matched Filter (CMF) provided however that possible effects of the AGC are eliminated. This measurement is carried out also at the GW (single measurement per beam/satellite).

RSCP (Received Signal Code Power)
This measurement is carried out on both Primary and Traffic Channels of both MTs and on the channels received at the GW. Reference point for the measurement is either the output of the on-time code correlator or of the rake combiner in case of diversity operation. The measurement cannot simply consist in the evaluation of the power at the reference point at least for the case of strongly coded signal. In that case in fact the operating SNR is close to zero or lower and the simple power measurement will include not only the signal power but also the residual noise and interference. A strong bias in the measurement will then result. To reduce the effects of residual noise and interference, where available use of the channel reference symbols can be made. In such cases, the channel estimator output signal power can be used.
In case reference symbols are not present in the channel considered for the measurements, more sophisticated algorithms are required. In particular, two options are available: use tentative, or final, data decisions to remove modulation (Decision Directed approach) or use a non-linear transformation to recover an unmodulated signal component, which can be used in place of the reference symbols. The last option is preferable for its simplicity although it is not necessarily best performing. Finally, in all cases, effects of AGC have to be eliminated for having meaningful results.

BLER (BLock Error rate)
The BLER measurement can be estimated via the CRC processing of each transport block. A trivial way to implement the measurement is through counting of transport blocks in error over a given time window. This approach for the measurement may require a quite long time for achieving reasonably low error rate measurement capability.
Such an approach is thus not suitable for real time procedures (like *Link Status* derivation). For such real time procedures simple indication of Block Error event may be sufficient.

Transmitted Power

This is a measurement concerning the current power setting of the MT and GW transmitters.

IV. A T-S-UMTS Core Network

Two different approaches of an integrated T-S-UMTS CN are possible, the two-domain case and a single domain case. In the two-domain case the satellite access network together with its own CN are connected via an interworking function (IWF) or a GGSN (Gateway GPRS Support Node) to a separated T-UMTS CN that is run by another operator. In the single domain case the S-UMTS segment is directly connected via an SGSN (Serving GPRS Support Node) to the same CN where the T-UMTS segment is connected to as shown in Figure 4. As for VIRTUOUS the single domain case has already been chosen in FUTURE only the single domain case will be taken into account.

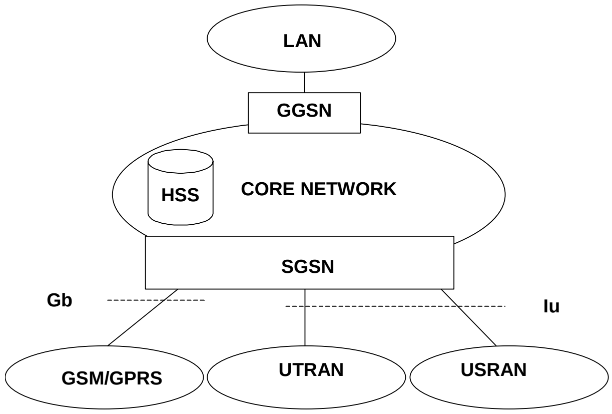

Figure 4: Single Domain Case

For the satellite segment it can be assumed that only a few ground stations will exist, e.g. IRIDIUM has 21 earth stations worldwide[10]. Thus, on one hand the routing within the CN during the call will be against zero and on the other hand the SGSN connected with the satellite segment should have enough bandwidth to

serve all satellite connections in an adequate way, which comprises especially the delay characteristic. This means that the satellite segment can be connected to a conventional GPRS CN by an SGSN as far as enough resources are available. Thus to guarantee QoS in the CN resource reservation might be useful, e.g. using RSVP (Resource Reservation Protocol).
No explicit in-call routing algorithms have to be implemented for the satellite in the CN; especially in a GEO constellation every type of routing can be avoided, e.g. by selecting the best CN operator in advance. In this case the connection will be directly established with the appropriate SGSN. Another possibility is to chose the groundstation due to QoS requirements. To achieve this a central CAC mechanism has to be established who has knowledge about the connection parameter of all accessible groundstations and the affiliate SGSNs/CNs.

V. Conclusion and Outlook

The FUTURE project has made an approach to achieve end-to-end QoS in a packet based integrated T-S-UMTS network focusing on the satellite based UTMS radio access network.
Three different algorithms concerning the USRAN have been investigated with the support of a measurement control unit. All these algorithms will be tested in the FUTURE test bed which is a further development of the VIRTUOUS demonstrator. As the FUTURE QoS concept is embedded in the UMTS QoS concept and architecture all the algorithms presented in the paper will be compatible with the approaches for the packet support over the terrestrial UMTS networks.
It is assumed that the be an important contribution to the end-to-end QoS achievement in packet oriented multimedia environment, which will be made ubiquitous by the integration of an satellite segment.

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