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Speech Quality Aware Admission Control for Fixed IEEE 802.16 Wireless MAN

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***Abstract*—We present an Admission Control (AC) algorithm for the specific case of pre-provisioned IEEE 802.16d links for VoIP aggregates. The algorithm approaches AC from a new perspective as admission criterion is speech quality, the sole true quality metric for voice services. As we found the E-Model can be used to reliably estimate speech quality and the resulting R-Score as admission criterion. Moreover, we show that speech quality can be evaluated on aggregate level without compromising individual call's speech quality. The algorithm is simple, fast, and precise and its behaviour is consistent over a range of different deployments.**

I. Introduction

With the release of the IEEE 802.16 Broadband Wireless MAN standard family (IEEE 802.16d/e [1], [2]), a powerful technology is seizing dominance in Broadband Wireless Access (BWA). The foundation of the WiMAX (Worldwide Interoperability for Microwave Access) Forum, an "industry-led, not-for-profit organisation formed to certify and promote the compatibility and interoperability of broadband wireless products based upon the harmonised IEEE 802.16/ETSI HiperMAN standard" [3], is certainly a reliable indicator therefore.

This belief in the potential of IEEE 802.16 has enlarged in the research community too and much research efforts are underway as there is still a lack of understanding and tools [4]. This quest for insight is further stimulated by the decision of the IEEE 802.16 Working Group to leave crucial parts, especially related to performance, and largely unspecified in order to provide manufactures with a powerful tool to distinguish their products.

Among many others, one of these functions is Admission Control (AC) as neither of both standards, IEEE 802.16d for fixed BWA, nor in 802.16e, the amendment for mobile scenarios, specify any AC mechanism. Notwithstanding, IEEE 802.16d defines a comprehensive Quality of Service (QoS) model, which itself depends on AC.

The lack of AC was the motivation for this work and in this paper we present a tailored Measurement Based Admission Control (MBAC) algorithm for the specific case of Voice over IP (VoIP) services delivered over pre-provisioned IEEE 802.16d links. The presented algorithm approaches the AC problem from a different perspective and is based on a cross-layer design. As metric for quality of VoIP services we apply Subjective QoS (SQ) assessment based on speech quality. It is evaluated at application layer by Objective QoS Assessment (OQA), which in turn builds on statistics from the packet loss process, captured on MAC layer. Finally, QoS control in from of AC is applied on MAC and IP layer.

The paper is structured as follows. First we review relevant IEEE 802.16 details and the envisioned deployment context, altogether in Sec. II. In Sec. III we briefly review general principles of AC before we introduce the OQA method used to evaluate the admission criterion. Thereafter, in Sec. IV we present and discuss the performance results revealed by simulation before we close the paper with a conclusion in Sec. V.

II. IEEE 802.16d Background and the Deployment Scenario

The IEEE 802.16d standard, officially called 802.16-2004 with reference to its release date, defines an air interface for fixed BWA. In doing so it specifies several physical layers (PHY) and a Common MAC (MAC) layer on top of them. Target deployment is fixed Non-Line-of-Sight (NLOS) within the 2GHz to 11GHz frequency band, either in Point-to-MultiPoint (PMP) or in mesh mode. In PMP, a central Base Station (BS) controls all traffic interactions between Subscriber Stations (SS) and itself and all traffic is sent from a single SS to the BS, called Uplink (UL) or from the BS to one or many SSs, called Downlink (DL).

The MAC is connection oriented in order to support QoS, an essential future BWA requirement. There are several types of connections, each unidirectional and between two MAC instances. These connections serve different purposes like MAC management and signalling with several priorities but also for data transport. Connections are identified by a unique Connection Identifier (CID).

Besides connections, IEEE 802.16d defines the concept of Service Flows (SF). An SF itself is defined as unidirectional transport service with predetermined QoS characteristics, i.e. QoS parameters. Each SF is mapped to a single connection and has to be served by an UL (or DL) scheduler such that QoS requirements are met. This is being done by a so-called scheduling service which is related to the QoS parameters associated with the respective SF. It should be noted that the standard defines scheduling services but does not define any explicit scheduler for them. It is left to manufactures to select and implement a scheduler which meets the respective requirements. In this respect, IEEE 802.16d is in line with concepts known from DiffServ, which specifies Per Hop Behaviours (PHB), see for example [5], but not how to implement them.

One of the envisioned deployments of IEEE 802.16d is to deliver VoIP services in different granularities. As IEEE 802.16 is connection oriented the spectrum ranges from single VoIP call to VoIP aggregates. We focus on aggregates as single VoIP calls are rather typical for scenarios involving mobile terminals. As this work is in the context of the European research project "WiMAX Extensions for Isolated Research Data Networks" (WEIRD) [6], the deployment we have in mind is a real deployment scenario defined by WEIRD. In this scenario a remote monitoring station, in the role of an SS, is connected by a BS to a central unit. In reality this is a Forest Fire Monitoring Station (FFMS) somewhere in the mountains connected to a Coordination Centre (CC) in a nearby city. In this case VoIP services are used to support the personnel in the FFMS in reporting and for coordination of forest fire prevention activities by the CC. In order to do so, this scenario defines a dedicated, pre-provisioned SF with a certain, fixed capacity in either direction. For more details on this scenario we point the readers to [7]. The scenario is illustrated in Fig. 1.

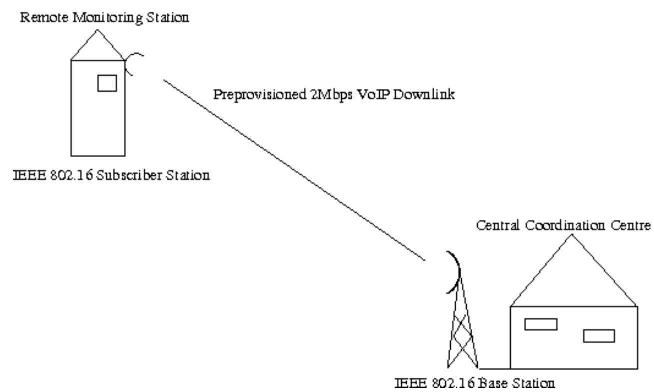

Fig. 1. The example deployment scenario as defined in the EU IST FP6 IP Project "WEIRD". A Remote Monitoring Station (Forest Fire Monitoring Station) is connected with a Coordination Centre via a pre-provisioned IEEE 802.16 Service Flow for VoIP services.

III. The Principle of Admission Control and its Application to VoIP

Admission Control is the most important mechanism for QoS provisioning on aggregate level. In other words, if a provider decides to exploit statistical multiplexing gain within a single traffic class, AC regulates the traffic intensity by controlling the number of active flows such that a certain QoS objective is met. In the context of our VoIP scenario, this means an AC function controls the VoIP traffic arriving at the aggregation point, i.e. the BS in DL direction or the SS in UL direction and destined to either the FFMS or the CC, such that a certain VoIP quality is assured.

Derived from the definition presented in [8], AC can be generally defined as

$$\chi_k \begin{cases} \geq 0 & \text{admit flow k} \\ = 0 & \text{reject flow k} \end{cases} \quad (1)$$

where χ_k denotes the admission criterion for the requesting flow k and is defined as

$$\chi_k = \max\{Q(N+1) - Q', 0\}. \quad (2)$$

Here we assume that $Q(n)$ expresses the level of QoS for n admitted sources, i.e. the traffic aggregate, and is a monotonically decreasing function in n, while Q' is the target QoS. The computation of $Q(N+1)$ reads as

$$Q(N+1) = Q(N) - \Delta_{QoS}^{K} \quad (3)$$

where Δ_{QoS}^{K} denotes the QoS degradation inflicted on the aggregate by the characteristics of flow k, if the latter would be accepted.

A. Objective Speech Quality Assessment using the E-Model

As mentioned in Sec. I, we approach the AC problem from a different (novel) perspective, i.e. based on speech quality determined by OQA. But before we go in details, we briefly describe our motivation for this decision.

If scientists speak about QoS they frequently imply what is known as Intrinsic QoS (IQ), cf. [9]. A few examples in our context are [10]–[12]. Intrinsic QoS means that QoS level is expressed by standard physical network parameters like delay, loss and jitter and in general by the expectation of the latters. The obvious reason is that IQ can be easily measured and has an intuitive, precise meaning. A likely mismatch with user perceived QoS at the user interface, called Subjective QoS (SQ), is in general silently ignored. In fact, in particular for VoIP services quality means intelligibility and therefore has to be evaluated on a higher, application layer and based on speech quality [13]. Intuitively, for interactive services involving human perception, subjective rating is the ultimate measure. Thus, the question is how to assess user perceived quality.

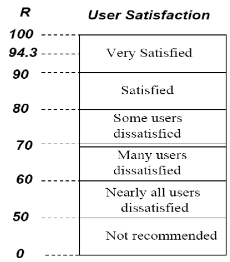

Fig. 2. Mapping Average User Satisfaction (Mean Opinion Score, MOS) to the R-Score.

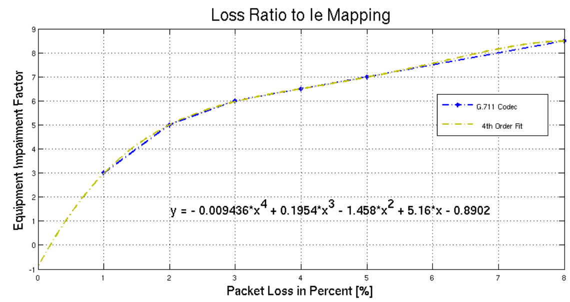

Fig. 3. Non-linear relation between the Packet Loss Ratio and the Equipment Impairment Factore (I_e).

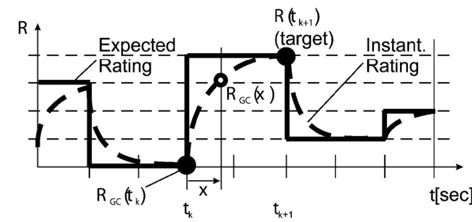

Fig. 4. A series of consecutive periods of different microscopic loss behaviours, i.e. packet loss ratio and distribution, together form a macroscopic loss profile. The pictures shows the expected rating (solid line) associated with either loss or gap state. It also indicates the true, delayed perception (dashed line) by humans as an exponential decay or rise of the R-Score with respect to a state transition. Picture source: [17].

There are several methods to assess speech quality for VoIP, see [13] for a complete treatment of this topic. Many of them involve surveying humans and are called *auditory* or *subjective* methods and correspond to SQ. Obviously, these methods are not suitable for systems and for this purpose, so-called *objective* or *instrumental* methods have been developed. Instrumental methods correspond to what we call OQA and are located in between IQ and SQ as they derive SQ ratings from measured network parameters.

The de-facto objective method is the E-Model. Started as a study by the ETSI, it has been standardised by the ITU-T [14]. Its original application domain is network planning and one of the questions we try to answer in this paper is if it lends itself for online resource management. The E-Model is a method for objective mouth-to-ear transmission quality assessment based on human perception and is defined as

$$R = R_0 - I_s - I_d - I_e + A \tag{4}$$

In (4), R denotes the psychoacoustic quality score defined in [0, 100]. It is an additive, non linear quality metric based on a set of impairment factors. Noise and loudness effects are represented by R_0, while I_s denotes speech signal impairment like for example PCM quantising distortion. Both are intrinsic to speech signal processing itself. Impairment imposed by transport is represented by I_d, which stands for speech signal delay impairment and I_e for equipment such as IP networks. Eventually, A is the advantage factor, a compensator for poor quality along with a convenience gain (e.g. cell phones). To assess the quality of a VoIP call, one has to compute and add the individual components of (4). The relation between R and human satisfaction, expressed by the Mean Opinion Score (MOS), is a result of extensive auditory tests and can be found in [14]. It is depicted in Fig. 2.

B. Instationary Quality Distortion and Human Perception

Speech quality is mostly a function of the packet loss ratio, i.e. I_e [13]. As with the relation of R and MOS, this relation has been found by extensive auditory tests. It is non-linear and the relevant part for our purpose is depicted in Fig. 3. Using a simple 4th order least square fit the function reads

$$I_e = -0.009436x^4 + 0.1954x^3 - 1.458x^2 + 5.16x - 0.8902. \tag{5}$$

But measuring packet loss and mapping it to I_e, as for example in [15], is insufficient as speech quality is further determined by packet loss distribution. Intuitively, single packet losses are always preferable over loss bursts. Exactly this makes the difference between IQ and SQ since by taking averages, as with IQ, such details are ignored. Furthermore, packet loss distribution itself is rather instationary over a call's life time and instantaneous as well as ultimate quality rating by humans exhibits strong correlation with this characteristic [13, Chap. 4].

To account for this phenomenon we divide the packet loss process in periods with different loss behaviours, as proposed by [16] and refined in [13], [17], [18]. In particular, we adopt the principles of the model proposed by Clark [16] but modified it for our purpose. Essentially, this packet loss driven model defines two alternating states, loss gap and loss burst state, with respect to the distance of packet loss events, cf. Fig. 4. As long as there is a minimum of 16 successfully received packets between two loss events, the model remains in (loss) gap state, otherwise there is a transition from gap to (loss) burst. The idea behind staying in gap state under this condition is that modern VoIP codecs can handle isolated packet loss. In case of a transition to burst state, the model remains in this state until 16 packets were successfully received between the latest and the previous loss event.

At the detection of any state transition the loss ratio for the previous state is used to calculate the corresponding impairment level, I_e, using the relation depicted in Fig. 3, resulting in a time series of I_e values with respect to states.

But before these values can be used to compute R, there is another feature, inherent to human perception, which has been integrated in this model, the delayed perception (or acceptance) of quality change.

Naturally, humans tend to perceive a quality change rather continuously and not instantaneously at a state transition. Furthermore, there is a difference from good to bad and vice versa. So do humans, for example, confirm a change from good to bad much faster than the other way around. Generally, this feature can be modelled by an Exponential function, similar to a transistor saturation curve, with specific time constants. Given $I_{e,g}$ and $I_{e,b}$, the impairment linked to gap or burst, I_1 is the estimated instantaneous impairment level at the change from burst to gap condition and I_2 equals the level at the return from gap to burst. In mathematical terms, I_1 and I_2 can be expressed as

$$I_1 = I_{e,b} - (I_{e,b} - I_2)e^{-b/\tau_1} \tag{6}$$

$$I_2 = I_{e,g} + (I_1 - I_{e,g})e^{-g/\tau_2} \tag{7}$$

Here g and b denote the sojourn time in gap or burst state and τ_1 and τ_2 are the time constants, respectively. Typical values are $\tau_1 = 9s$ and $\tau_2 = 22s$ [13]. Proper combination of (6) and (7) yields an expression for I_2 independent from I_1:

$$I_2 = I_{e,g}(1 - e^{-g/\tau_2}) + I_{e,b}(1 - e^{-b/\tau_1})e^{-g/\tau_2} \tag{8}$$

Using (8) we now can calculate the average impairment level over a certain time, e.g. for the life time of a call. Therefore we first calculate average gap and burst length, $\bar{b}$ and $\bar{g}$, as well as the average impairment levels $\bar{I}_{e,g}$ and $\bar{I}_{e,b}$. Putting these in (8) and integrating it over one burst and gap yields the average impairment level for a certain loss profile of certain length. It reads:

$$\begin{aligned}\bar{I}_e = \frac{1}{\bar{b}+\bar{g}} * [\bar{I}_{e,b} * \bar{b} + \bar{I}_{e,g} * \bar{g} + \tau_1 * (\bar{I}_{e,b} - I_2) \\ *(e^{\bar{b}/\tau_1} - 1) - \tau_2 * (\bar{I}_{e,b} - (\bar{I}_{e,b} - I_2) \\ *e^{-\bar{b}/\tau_1} - \bar{I}_{e,g}) * (e^{\bar{g}/\tau_2} - 1)].\end{aligned} \tag{9}$$

Eventually, by replacing I_e in (4) with $\bar{I}_e$ and using proper values for the remaining parameters, one can evaluate the subjective quality for a single call by this instrumental method called *Integral Quality by Time Averaging* [13].

C. Admission Control based on Objective QoS Assessment

Equipped with the expressions derived in the previous sections we can formulate an admission criterion based on speech quality. Therefore we replace $Q(N+1)$ in (2) with $R(R_0, I_s, I_d, I_e, A)$ and set $R_0 - I_s = 94$, the default value with respect to inherent features of the G.711 codec. Further, I_d is set to an upper bound determined by the buffer length ω and the link capacity C, see [19] for details. Beyond this bound, packet delay translates into packet loss and is captured by I_e. The respective equation for I_d reads:

$$I_d = 4 + 1 * (\omega/C) \tag{10}$$

Combining all pieces and further assuming the worst case, i.e. we set A to zero, we get

$$\bar{R}_T = 94 - 4 + 1 * (\omega/C) - \bar{I}_e(T) \tag{11}$$

In this equation parameter T in $\bar{I}_e(T)$ indicates that the average impairment factor for time-varying speech quality assessment has been calculated over a window of T seconds. This is to account for an inherent feature of Measurement Based AC (MBAC) algorithms, which always estimate a QoS criterion over a limited window. Eventually, we can express the admission criterion as follows:

$$\chi_k = \max\{\bar{R}_T - R', 0\}. \tag{12}$$

It has to be noticed that the criterion in (12) slightly differs from the one in (2) as we put Q(N) ($\bar{R}_T$) in place of Q(N+1). This is due to the difficulty in expressing and quantifying Δ^K_{QoS} without a precise traffic model. As we will show this is of little or no impact but we currently investigate alternatives and their merit.

Furthermore, by using this setup speech quality is assessed on aggregate level with a method that is originally designed to assess individual call quality. If this makes sense at all is discussed in the sequel, see Sec. IV-B2. From a model point of view, however, there is little difference in computing $\bar{I}_e$ on aggregate or call level. What is needed in both cases is loss ratio, burst and gap length. The only difference is the number of packets received (or lost) to trigger state transition, which is 16 for a single call, cf. Sec. III-A.

In order to translate this trigger threshold to aggregate level we apply a simple, intuitive approach. The AC algorithm knows at any time the number of admitted flows N. By assuming that VoIP traffic can be modelled by a standard Exponential On/Off model with an average sojourn time in On (talk) state of 300ms and mean Off (silence) 600ms [18], we know that each flow is active (On) for roughly on third of its life time. On the basis of this, we set the number of packets received (or lost) to trigger state transitions to 16*N*0.33.

IV. Performance Evaluation

A. Experimental Setup and Parameterisation

In order to evaluate the concept and performance of the algorithm, we implemented it in the NS-2 framework [20]. The basic scenario has been already described in Sec. II and complies with an evaluation scenario defined by WEIRD. In this scenario a pre-provisioned SF is set for VoIP. By definition this implies a contracted and assured capacity at any time and if there is any channel instability, it is compensated by scheduling decisions or any other mechanism. We further assume no packet loss over the air interface by using retransmissions or simply by appropriate network planning. In such a scenario the UL and DL AC function, placed in the SS or BS are equivalent. It means that we can reduce the simulation setup to a single server queue with fixed capacity.

The pre-provisioned link capacity of the respective SF, called *Minimum Reserved Traffic Rate* in IEEE 802.16 QoS

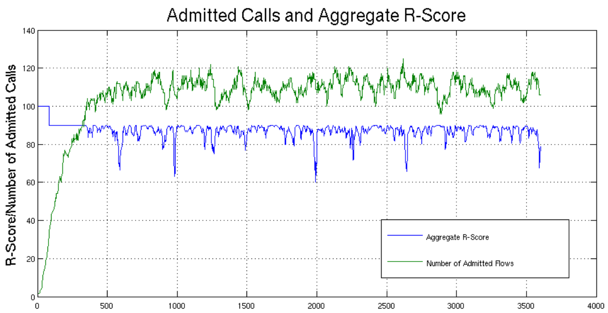

Fig. 5. The figure shows the number of admitted flows (upper curve) at the time of an admission request. At that time, the aggregate R-Score is estimated (lover curve) which serves as admission criterion. In this simulation R' in (11) was set to 85 (MOS: Satisfied). As indicated, this target has been closely achieved over the time in steady-state.

TABLE I
EVALUATION OF THE AC ALGORITHM'S PRECISION WITH RESPECT TO R'

R'	$\bar{R}_{T,\mu}$	R_σ	R_{min}	R_{max}	$t^{max}_{R<R'}$
80	84.67	4.44	63.36	89.94	31.73
82	85.55	4.16	68.37	89.95	29.66
84	86.53	3.57	66.04	89.98	28.12
86	87.64	2.77	67.05	89.97	37.76
88	88.63	1.71	77.88	89.97	43.22

terminology, has been set to 2Mbps and the buffer has a length of 30 packets. Call arrivals follow a Poisson process with mean arrival time of 2s and the holding time is exponentially distributed with mean 210s.

Voice over IP traffic was generated by a G.711 coder with voice fames of 20ms length. The standard Exponential On/Off model is used to model talk and silent periods where average sojourn time in On state is 300ms and mean Off 600ms [18].

Admission control is implicit and new calls are detected at the first packet arrival. The algorithm's window length, the past time over which speech quality is assessed, is set to 300s in order to cover a call of average length.

Finally, all simulations run for 3600 simulated seconds and the first 500s are discarded to evaluate the system in steady-state.

B. Performance Results

1) Admission Control Accuracy: One of the fundamental problems of MBAC is precision and only a few algorithms tackle this issue [21]. Hence, we first investigate how closely the algorithm approaches a demanded QoS target.

For the first simulation R' in (11) was set to 85 and as shown in Fig. 5, this target was achieved for most of the time. Skipping transient state the average estimated R-Score ($\bar{R}_{T,\mu}$) for the remaining time was 86.82, standard deviation $R_\sigma = 3.89$, $R_{min} = 59.97$ and $R_{max} = 98.89$. Additionally, we computed the longest continuous period below R', $t^{max}_{\bar{R}_T<R'}$ and found a value of 39.87s. We repeated this simulation for different R' in the range [80, 90], which maps on MOS to "Satisfied". The results are listed in Tab. I.

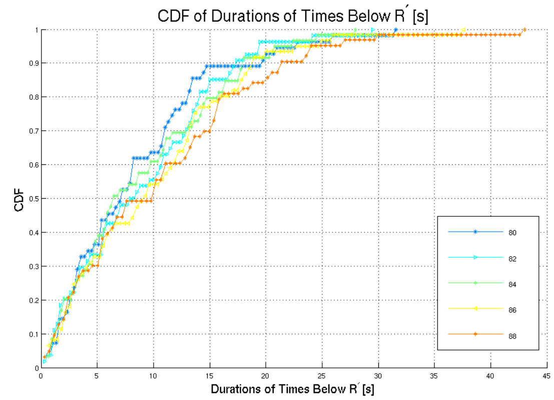

Fig. 6. Cumulative Distribution Function (CDF) of $t_{\bar{R}_T<R'}$. The probability that $t_{\bar{R}_T<R'}$ is larger than $1/10$ of the holding time, in other words that the quality is below the requested one for one tenth of a calls life time, is approximately 0.2.

The results indicate a relative consistent performance but the AC appears a bit too conservative for lower R' values. Perhaps more important in the context of traffic aggregates and statistical QoS is that the average R-Score was slightly above R' in all simulations. Among the remaining parameters, $t^{max}_{R<R'}$ certainly holds the most interesting information. At first sight the maximum duration seems relatively large compared to an average holding time of 210s. But the maximum alone does not tell much and in Fig. 6 we plot the CDF of the times $\bar{R}_T$ remained below R', denoted by $t_{\bar{R}_T<R'}$.

This figure further indicates consistency as the curves are very similar. For the whole range of R' values the average time $\bar{R}_T$ is below R' is approximately around 10s and the probability that $t_{\bar{R}_T<R'}$ is larger than 20s is around 0.2. This qualifies the large value for $t^{max}_{R<R'}$.

2) Subjective QoS Performance on Call Level: Speaking in general terms, what has been achieved by now is an algorithm that can statistically guarantee a predefined application layer metric. But how meaningful is this metric on call level? Can we assume that an R-Score measured and maintained on aggregate level applies to individual calls too?

In order to find this out we ran the same set of simulations as before, selected randomly around 100 consecutively admitted flows and recorded their loss process. We then used the same OQA with an adjusted state transition trigger, see Sec. IV-A. The question we tried to answer is how many of these flows receive the contracted QoS.. Hence, we assessed SQ for each flows total life time and Fig. 7 plots the CDF of these calls R-Scores. The figure shows that for each QoS target R' maximally around 5 percent of calls are rated below R=80, which is the lower threshold for "Satisfied" on MOS scale, cf. Fig. 2. Taking the first simulation, depicted in Fig. 5, as example it means that approximately 6 flows out of 110 concurrently admitted flows in average would be affected by lower QoS than contracted. However, some of them fall still in the range R=[70, 80] which maps to MOS "Some Users Dissatisfied", meaning that some of these may still rating

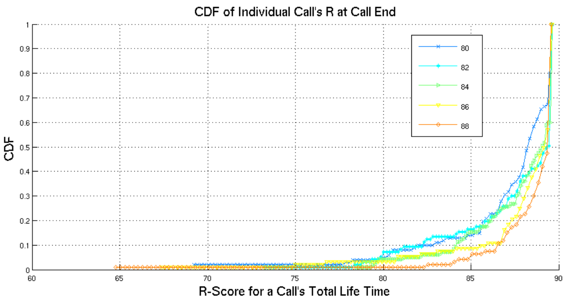

Fig. 7. Cumulative Distribution Function (CDF) of single call quality for a set of randomly recorded calls for all simulations. For each simulation less than 5 percent of calls fall below R=80.

"Satisfied".

Finally, from Fig. 5 we can draw conclusions with respect to configuration and QoS vs. resource utilisation trade-off.. If a operator wants to make sure that approximately less than 2 percent of calls fall below R=80 (MOS: Satisfied), it should set $R' = 88$. Obviously, the higher the QoS demands, the less the network utilisation. Hence, an operator can trade-off between user satisfaction and resource utilisation. It appears to us, that a in our setup the configuration $R' = 84$ seems the best trade-off as there are around 5 percent below MOS "Satisfied" while roughly half of them are still in the range of MOS "Some Users Dissatisfied".

V. Conclusion

The conclusions of this work are manifold. On top of the list we found that the E-Model lends itself as metric for QoS control by MBAC. The necessary computations are simple don't add much burden on equipment. This opens the door to a new domain in VoIP QoS control, namely based speech quality, the only reliable QoS assessment method for VoIP. In support of this statement, we found that the algorithm exhibits a consistent and accurate behaviour for a whole range of configurations. Probably the most intriguing conclusion and somewhat specific to our setting is, that with a slightly modified measurement procedure, we could apply the model on aggregate level without compromising call level speech quality.

But there is still room for further refinement. So do we currently investigate, as already mentioned in Sec. III-C, ways to express Δ^K_{QoS}. Finally, only a comparison with common IQ based MBAC algorithms prove the algorithms superiority or not. It is in preparation and will be published in a follow-up.

Acknowledgement

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