Performance Optimization of Single-Cell Voice over WiFi Communications Using Quantitative Cross-Layering Analysis

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Abstract. Cross-layer design has been proposed to optimize the performance of networks by exploiting the inter-relation among parameters and procedures at different levels of the protocol stack. This may be particularly beneficial in wireless scenarios, and for quality-of-service support. This paper proposes a quantitative study of cross-layer performance optimization for Voice over WiFi communications, which enables design engineers to analyze and quantify inter-layer dependencies and to identify the optimal operating point of the system, by using cost-benefit principles. Furthermore, insight gained on the problem enables the proposal of design principles for a Call Admission Control scheme able to enhance the overall system performance by limiting the number of users in the system and signalling to the active terminals of the proper parameter settings to optimize overall performance.

Keywords: Cross-Layer design, metamodeling, VoIP over WiFi.

1 Introduction

The layering principle has been long identified as a way to increase the interoperability and to improve the design of telecommunication protocols, where each layer offers services to adjacent upper layers and requires functionalities from adjacent lower ones. Standardization of such protocol stacks in the past enabled fast development of interoperable systems, but at the same time limited the performance of the overall architecture, due to the lack of coordination among layers. This issue is particularly relevant for wireless networks, where the very physical nature of the transmission medium introduces several performance limitations (including time-varying behavior, limited bandwidth, severe interference and propagation environments) and thus, severely limits the performance of protocols (e.g. TCP/IP) designed for wired networks.

To overcome these limitations, a modification of the layering paradigm has been proposed, namely, *cross-layer design*, or “cross-layering.” The core idea is to

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maintain the functionalities associated to the original layers but to allow coordination, interaction and joint optimization of protocols crossing different layers.

Several cross-layering approaches have been proposed in the literature so far [1, 2, 3, 4]. Nevertheless, little formal characterization of the cross-layer interaction among different levels of the protocol stack is available yet, with the exception of [5], where the impact of different layers is studied in order to optimize service delivery in mobile ad-hoc networks, and [6], where the authors introduced a meta-modeling approach to study cross-layer scheduling in wireless local area networks.

A clear need is emerging for identifying approaches able to analyze and provide *quantitative guidelines* for the design of cross-layer solutions, and, even more important, to decide whether cross-layering represents an effective solution or not. In [7], we initiated a quantitative approach for calculating the sensitivity of system performance with respect to parameters across different layers for a simple Voice over WiFi system.

Voice over WiFi (VoWiFi) communications represent a challenging scenario, as, even in the simplest case of a single IEEE 802.11 cell, performance optimization requires the consideration of several parameters at different levels of the protocol stack. Indeed, codec parameters as well as link layer and physical parameters (and several others) clearly have impact on the overall quality of communication as it is perceived by the end user.

As limited quality-of-service strategies are employed on the wireless link, there is a need for a proper *Call Admission Control* (CAC) strategy [8-16], in order to provide ways to limit the number of users in the system and, more generally, to provide possible on-line adjustments to terminal parameters.

In view of the above, this paper describes the use of a formal framework to (1) identify and formalize the interactions crossing the layers of the standardized protocol stack; (2) systematically study cross-layer effects in terms of quantitative models; (3) support the design of cross-layering techniques for optimizing network performance; (4) define design principles of a CAC strategy for the system to work at, or near to, its optimal operating point. The presented approach, based on techniques well-established in operations research, allow engineers to identify correlations among different parameters and to estimate the potential advantages (if any) deriving from cross-layer interactions.

The structure of the paper is the following: Section 2 summarizes our formalization of parameters and measurements at different layers of the protocol stack in layering and cross-layering schemes. Section 3 describes a specific VoWiFi single-cell scenario, cross-layer signaling implementation and the process of system modeling. Furthermore, the section discusses a formal *cost-benefit analysis* to optimize the performance of the system from the service provider and the wireless terminal perspectives. Section 4 derives design principles of a candidate CAC mechanism to optimize the overall performance and, finally, Section 5 draws conclusions and outlines future work on the topic.

2 Cross-Layer Design and Cost-Benefit Analysis

In a previous paper [7] we outlined a formal framework for cross-layer design by identifying system parameter vectors as the merging of two sub-arrays representing, respectively, internal and external parameter subsets: $\overline{p}^N = \left[\overline{p}_i^N \mid \overline{p}_e^N\right]$, $\overline{m}^N = \left[\overline{m}_i^N \mid \overline{m}_e^N\right]$. Cross-layer design allows a large degree of flexibility, by enabling a higher level of interaction among the entities at any layer of the protocol stack. Layer N is enabled to control, depending on the specifics, a subset of all the parameters at any level $\overline{p}^{TOT} = \left[\overline{p}^1 \mid \overline{p}^2 \mid ... \mid \overline{p}^7\right]$ and can acquire measurements as a subset of $\overline{m}^{TOT} = \left[\overline{m}^1 \mid \overline{m}^2 \mid ... \mid \overline{m}^7\right]$ where we assume an OSI-like seven layer stack.

Cross-layer design derives from the observation that the performance of a network or other system depends on several mechanisms situated at different levels of the protocol stack interacting in a complex fashion. *Quantifying* the effect of these interactions is very important in order to be able to systematically relate such interactions to system outcomes and be able to quantify the decision to take such interactions into account – using a cost-benefit analysis, so that the benefits outweigh the cost of additional complexity and "layer violation" [17, 7].

In [7] we advocated the use of formal system modeling to express cross-layer interactions and their effect on system performance, based on sensitivity analysis. The system response with respect to the k-th performance metric is modeled as a function jointly of all parameters across the layers, $f_k()$. The sensitivity of the system response and the interactions among factors, within and across layers, can then be captured naturally as the partial derivatives $\frac{\partial f_k}{\partial p_i^j}$ and $\frac{\partial^2 f_k}{\partial p_i^j \partial p_l^m}$, where p_i^j is the i-th parameter at layer j. Subsequently, one can then strictly or nearly optimize the performance e_i with respect to a subset of p^{TOT} under general constraints by using any available method, such as steepest ascent, stochastic approximation, ridge analysis, and stationary points [18, 19].

The function $f_k()$ across the layers can be analytically calculated or empirically estimated. Since closed form mathematical expressions are often unattainable for real systems, in [7] we outlined a mathematical modeling procedure based on *metamodeling*. In this paper, we continue and extend our work on metamodeling of wireless systems, by (meta)modeling the performance of a multi-user VoWiFi system with several parameters, and across several layers.

Our "raw" performance metrics, e_i, are further incorporated into a utility or "benefit" function $U(e)$ that expresses how valuable the (net) system performance is to the system owner or user. In general, the exact functional form of the utility and resulting objective function are less important than their curvature (often convex, to denote a certain "saturation") and their ability to preserve a relative ordering of the engineering alternatives, to enable ultimate design decisions.

Results achieved during the system optimization phase are then employed to define guidelines for system design. By employing the proposed framework, it is possible to select 1) the sensitivity of the system utility with respect to individual parameters; 2) the optimal operating point of the system (direct consequence of the optimization process); 3) the proper cross-layer interactions to enable (based on sensitivity of the system); 4) and the proper signaling architecture to employ (allowing to identify the set of parameters and measurements to use).

3 A VoIP over WiFi Scenario

In this section we illustrate the application of the proposed modeling approach in a VoIP over WiFi setting. The model is built in a four-dimensional domain defined by a set of parameters considered crucial for the overall system performance, namely, physical bandwidth, link error rate, maximum number of link layer retransmissions, and VoIP frame generation interval. The chosen set of parameters is spread over several layers of the protocol stack, making it difficult to predict the optimal operation point using ad hoc or intuitive methods.

3.1 System Model

Network Model
The network model is shown in Fig. 1. The network is an infrastructure WLAN with one Access Point (AP) serving *N* client nodes. Each client node initiates a bidirectional VoIP call with the AP. As the result, there are *N* uplink and *N* downlink calls carried in the network simultaneously. For each call, we use the ITU G.711 64kbps codec [20] where frames are sent for transmission at regular time intervals.

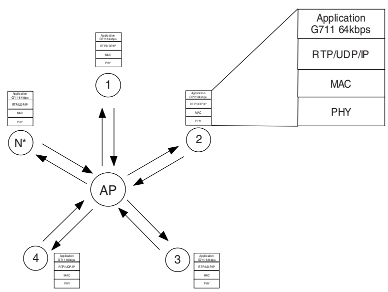

Fig. 1. Simulation scenario

The frames produced by the voice coder are then encapsulated by RTP/UDP/IP layers of the protocol stack adding an overhead of 40 bytes. In the MAC layer, IEEE 802.11 DCF basic access mode with no RTS/CTS exchange is used.

Inputs
The controllable or design variables of interest are the following:

Physical data rate (D) [1] is the data rate available for transmission at the physical layer. In order to comply with IEEE 802.11b, physical data rate values are taken equal to 1 Mbps, 2 Mbps, 5.5 Mbps and 11 Mbps.

[1] We acknowledge the fact that data rate D is determined by link adaptation algorithm on the basis of signal-noise ratio detected by the receiver. In the paper, we assume data rate can be controlled independently. The consideration of interaction of D with SNR will be exploited in future work with an appropriate link adaptation algorithm and accurate interference modeling at physical layer.

Packet Error Rate (PER). Wireless systems are usually characterized by a high error rate corrupting data transmitted at the physical layer. In fact, typical PERs for WLAN links are in the range between 10^{-3} and 10^{-1}. However, in order to evaluate system performance also for low error rate channels, we decided to vary PER between a lower value of 10^{-9} and 10^{-1}.

Maximum number of retransmissions (R). The task of link layer ARQ is to compensate high error rates on wireless channels. The crucial parameter for ARQ scheme performance is the maximum number of retransmission attempts performed before the link layer gives up and drops the frame. Each retransmission consumes the same physical resources as the original frame transmission, thus reducing the overall capacity of the cell. On the other hand, retransmissions increase packet delivery delay. In our network model, R is varied from 0 to 5, where 0 corresponds to the case when no retransmissions are performed at the link layer.

Voice packet interval (I) defines the time interval between frames generated by the voice codec. Voice packets are then encapsulated using RTP over UDP/IP protocols. Voice frames produced by the codec are relatively small (usually smaller than 100 bytes). As a result, a portion of network capacity is wasted on protocol overhead (40 bytes per packet). The parameter I is varied from 10 to 90 ms in our scenario.

Outputs

The output response of interest, $e=N^*$, is the maximum number of VoIP calls that can be supported by the WLAN cell with a satisfactory quality, which is defined by the following constraints.

Constraints

Several factors affecting VoIP performance can be mainly divided into *human* factors and *network* factors. Human factors define the perception of the voice quality by the end-user. The most widely accepted metric, called the Mean Opinion Score (MOS) [21], provides arithmetic mean of all individual scores, and can range from 1 (worst) to 5 (best).

The factors affecting the MOS ranking are related to network dynamics and include end-to-end propagation delay and frame loss [21, 22]. The delay includes the encoder's processing and packetization delay, queuing delay, channel access and propagation delay. For this reason, in order to ensure an acceptable VoIP quality, we limit the delay parameter to 100 ms measured between unpacketized voice data signal at codecs located at the sender and the receiver nodes. The second factor, frame loss rate, affects the VoIP quality due to non-ideal channel conditions. The chosen ITU G.711 64kbps codec [20] shows acceptable MOS rating (MOS=3) for frame loss rate up to 5% [23].

Cross layer Model of VoIP

Following from the above, we assume a quantitative model for the VoIP capacity as $N^* = f(D,PER,R,I)$ which we proceed to estimate via response surface (meta)modeling, since a closed form analytical model across the layers is clearly intractable.

3.2 Implementation and Cross-Layer Signaling

The network model is implemented in the ns2 network simulator (version 2.29) [24]. The simulation parameters are summarized in Table 1. The ITU G.711 64kbps codec [20] is emulated using Constant Bit Rate (CBR) generator source, producing blocks of data in regular intervals specified by the voice interval *I* input parameter. In addition to the voice codec, the Cross-Layer Control (CLC) module is added at the application layer of the protocol stack (see Fig.2). CLC is able to read the measured values of *D* and *PER* at the physical and link layers as well as internally *I* at the application layer. Moreover, it can set *R*, *I*, or *D* to the desired value.

Table 1. Simulation Parameters

Parameter Name	Value
Slot	20 µs
SIFS	10 µs
DIFS	50 µs
PLCP preamble + header	192 µs
Data Rate	1, 2, 5.5, or 11 Mb/s
Basic Data Rate	1 Mb/s
Propagation Model	two-ray ground
RTS/CTS	OFF

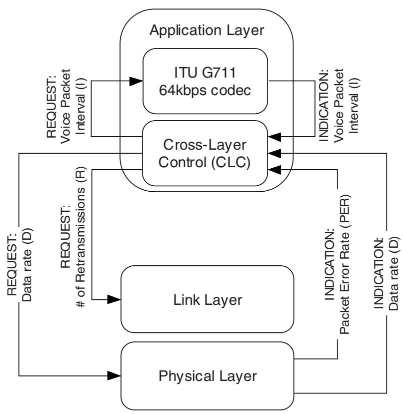

Fig. 2. Cross-Layer Control (CLC) module and cross-layer interactions

3.3 Model Definition

For each combination of input parameters, that is, *D*, *PER*, *R*, and *I*, we run a series of simulations with the number of VoIP flows incrementally set from 1 to 25. Then we find the maximum number of VoIP flows *N** accepted by the system as the output for which the quality of the voice signal remains above a satisfactory level (as defined in Section 3.1, with end-to-end delay less than 100 ms and frame error rate less than 5%), by checking every voice frame. Output capacity varies based on the controllable input variables change. Table 2 shows the values of input parameters used in the experiment.

In order to fit the simulation results with a model, we used the JMP [25] tool and a second order polynomial RSM model, with interactions which terms and corresponding coefficients are presented in the following equation (note that the interaction between *I* and *R* is not significant, therefore it is excluded from the model). Results show that R-square of the fitted model is equal to 0.81.

$$\begin{aligned} N^* &= -5.1027 + 1.5575 * D + 292.8806 * I + 1.3677 * R - 157.3738 * PER \\ &+ 5.9569 * D * I + 0.1980 * D * R - 5.1210 * D * PER - 891.6851 * I * PER \\ &+ 3.7706 * R * PER - 0.1186 * D^2 - 2710.813 * I^2 - 0.2935 * R^2 + 1644.7405 * PER^2 \end{aligned} \quad (1)$$

Table 2. Experiment Design Parameters

	Parameter Name	Abbreviation	Levels	Values
Inputs	Physical data rate	*D*	4	1, 2, 5.5, 11
	Packet Error Rate	*PER*	9	10^{-9}, 10^{-8}, 10^{-7}, 10^{-6}, 10^{-5}, 10^{-4}, 10^{-3}, 10^{-2}, 10^{-1}
	# of retransmissions	*R*	6	0, 1, 2, 3, 4, 5
	Voice packet interval	*I*	9	10, 20, 30, 40, 50, 60, 70, 80, 90
Constraints	Voice E2E delay	-	-	< 100 ms
	Frame error rate	-	-	< 5%

Fig. 3 illustrates the obtained metamodel *N** function in all four dimensions of *D*, *I*, *R*, and *PER*. The maximum of *N** with respect to *I* is located between 0.05 and 0.07 seconds at it is evident in Fig. 3a. Obviously, with the increase of *I*, client nodes generate fewer packets, thus increasing network capacity. However, the voice packet interval is included into the end-to-end delay constraint set to 100 ms. Consequently, after a certain threshold, an additional increase of *I* becomes unfavorable, leading to an overall network capacity decrease.

A similar observation can be made for the maximum number of retransmissions configured at the link layer. With a higher *R*, the system can sustain a higher error rate at the wireless link. However, each retransmission consumes bandwidth resources from the shared channel. For high data rate scenarios (11 Mb/s), retransmissions take just a small fraction entire bandwidth while for low data rate scenarios (1 or 2 Mb/s) the portion of bandwidth used for retransmissions becomes considerable (see Fig. 3b). As a result, the *N** is maximized at *R* equal to 3 for low data rates.

Fig. 3c illustrates that *N** is not sensitive with respect to low PERs (10^{-9} – 10^{-4}). However, when *PER* is high (> 10^{-4}) – which is often the case in WLAN networks – the system capacity dramatically decreases. The absolute maximum of *N** corresponds to *D=11 Mb/s*, *I=0.07 s*, *R=5*, $PER=10^{-9}$, and is equal to 20 bidirectional VoIP calls. The reader should note that this maximum corresponds to approximately 36% utilization of *D* provided at the physical layer. The remaining 64% is wasted on physical and link layer overhead which becomes especially relevant for small packets (like in VoIP). A detailed study of small packet performance under IEEE 802.11 WLAN as well as an optimization employing packet concatenation techniques is presented in [27].

Based on the above model, we proceed to quantify the sensitivity of the response $e=N^*$ on the four cross-layer variables D, I, R, and PER by calculating the derivatives:

$$dN^*/dD = -0.2372D + 5.9569I + 0.198R - 5.1209PER + 1.5575 \tag{2}$$

$$dN^*/dI = 5.9569D - 5421.626I - 891.6851PER + 292.8815 \tag{3}$$

$$dN^*/dR = 0.198D - 0.587R + 3.77PER + 1.3677 \tag{4}$$

$$dN^*/dPER = -5.12D - 891.6851I + 3.77R + 3289.48PER - 157.3774 \tag{5}$$

The knowledge of the behavior of the first-order derivatives of N^* allows the estimation of the impact of each of the parameters. The absolute maximum values of the derivatives are presented in Table 3. In our case, the voice packet interval I and packet error rate PER at the physical layer have a higher impact on the maximum number of calls N^* that can be supported by the system.

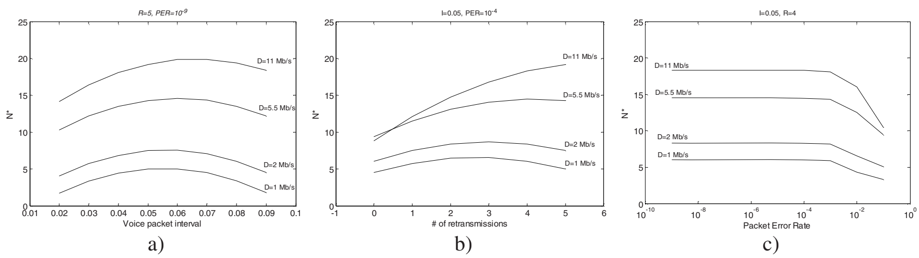

Fig. 3. Metamodel of the system VoIP call capacity (N^*)

Table 3. Absolute maximum values for N^* derivatives

Derivative	Maximum	D [Mb/s]	I [s]	R	PER
max $\lvert dN^*/dD \rvert$	2.84	1	0.09	5	0
max $\lvert dN^*/dI \rvert$	278.27	1	0.09	≥0	0.1
max $\lvert dN^*/dR \rvert$	3.92	11	≥0.02	0	0.1
max $\lvert dN^*/dPER \rvert$	293.95	11	0.09	0	0

3.4 System Optimization

Once the metamodel is established, it is possible to exploit the information it contains to build a utility or similar function and, thus, enable a cost-benefit analysis of the problem. In the case under examination, it is possible to identify two optimization scenarios, focused on the service provider and the wireless terminal, respectively, which are described in the following sub-sections.

Service Provider Perspective

From the point of view of the service provider, the main concern is associated with maximization of the profit obtained from the operating network. The profit is directly proportional to the number of calls that can be supported by the system simultaneously deducing network setup and operating costs.

Here, we define a utility function for the VoWiFi system as

$$U(D,I,R,PER) = N^{*} \cdot P_{call} - \frac{D_{wasted}}{D} \cdot N^{*} \cdot P_{call} - P_{power} \cdot (D_{norm} + PER_{norm})^{2} = \\ = N^{*} \cdot P_{call} \cdot (1 - \frac{D_{wasted}}{D}) - P_{power} \cdot (D_{norm} + PER_{norm})^{2} \tag{6}$$

where P_{call} is the price charged for (or the marginal income from) a single call, P_{power} is the marginal cost of a unit of transmitted power, and $D_{wasted} = D \cdot PER \cdot R / R_{\max}$ is the bandwidth wasted for re-transmissions, in packets/sec. The $N^{*} \cdot P_{call} \cdot D_{wasted} / D$ term accounts for the portion of bandwidth used for re-transmissions instead of voice traffic that the owner could have charged for. The last term is similar to the one used in [7] and captures the quadratic relationship of D and PER with respect to the radiated power.

Fig. 4 presents the behavior of U. The P_{call}/P_{power} ratio is chosen to be equal to 100 in our example. This corresponds to the policy of service provider to charge one dollar per VoIP call while the price paid for a power resource unit is just one cent.

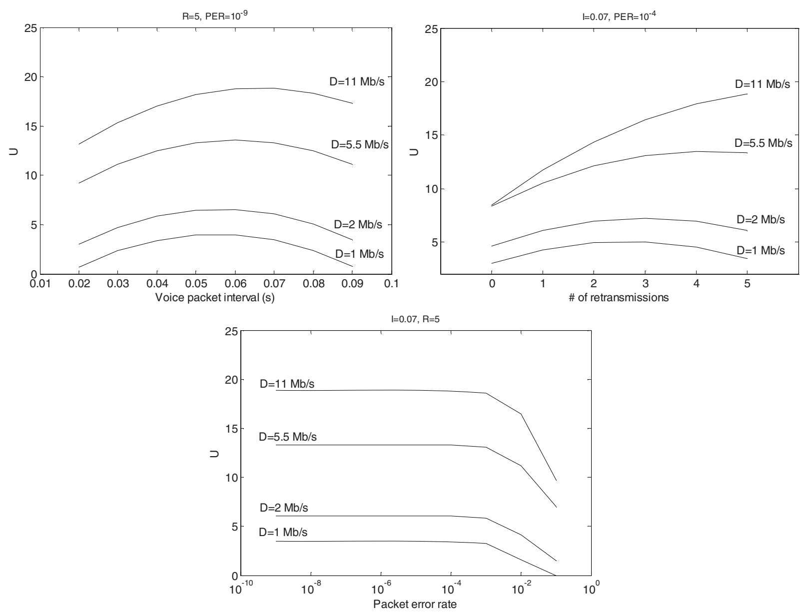

Fig. 4. VoIP network utility function *U* from the service provider point of view

The obtained utility function U is similar to the metamodel N^{*} describing VoIP system capacity. The max $U(D,I,R,PER)$ corresponds to the max of N^{*} at $D=11$ *Mb/s*, $I=0.07$ *s*, $R=5$, $PER=10^{-9}$ and is equal to 18.89 dollars. The difference with the maximum number of VoIP calls (max U) of one dollar is associated with the network operating costs.

Mobile Terminal Perspective

From the point of view of the mobile terminal, the main constraint can be identified in enabling long battery lifetime while providing acceptable voice performance. Since the latter is already included in the metamodel of the system, we can concentrate on the former to identify the cost related to the considered scenario. More specifically, the main parameters impacting on power consumption are the following:

- Transmission data rate *D*: in IEEE 802.11b, data rate selection at the physical level is based on the strength (or SNR) of the received signal. We use the same assumptions as in [26]: Power setting of Data Rate of 1Mbps = 30 mW; Power setting of Data Rate of 2Mbps = 35 mW; Power setting of Data Rate of 5.5 Mbps = 50 mW; Power setting of Data Rate of 11 Mbps = 100 mW. As a consequence, in order to support a given physical data rate *D*, the power should be adjusted to the appropriate value.
- Maximum number of retransmissions *R*: clearly, as the number of allowed retransmissions increases, more power is used for delivery of a single packet.

It is then possible to define a utility function to be maximized including such observations, where the relative weight of benefits against costs can be identify by varying the parameters α and β:

$$U = \alpha \cdot N^* - \beta \cdot \left[R + f(D)\right] \tag{7}$$

where the function $f(D) = 10^{1.4291 + 0.0515D}$ captures the above power constraint on receiver SNR in order to enable physical data rate *D*.

4 Design Principles for a VoWiFi Call Admission Control

The proposed framework represents a novel approach to cross-layer design and to the authors' knowledge it has not yet been addressed in the literature. Therefore, the purpose of this section is only to sketch a possible application scenario, beyond static performance optimization. In fact, on the basis of the analysis presented in the previous section, two empirical considerations are confirmed: (i) limitation on number of active nodes, and thus, an admission mechanism (CAC), is required in order to provide satisfactory performance to VoIP communications; (ii) the performance of the overall system significantly depends on several parameters, which can be recognized (and quantified) at different layers of the protocol stack, thus suggesting and justifying the use of a cross-layering scheme.

This motivates the introduction of a centralized call admission control to monitor the status of the overall VoIP system, which can exploit the metamodel information to provide the proper cross-layer parameter settings to perform run-time optimization of the system. Such CAC should be supported by the knowledge of the utility function (see Section 3) and could be implemented at the AP as the "central" point of the cell where all traffic converges. A example scenario could be the following:

– a new VoIP call is activated by a terminal;
– the CAC process on AP checks whether the cell already reached the maximum number of calls which can be supported (information inferred from the metamodel);

– if yes, the call is rejected;
– if no (i.e., there is room for an additional call), the CAC computes the optimal configuration of the cell to provide the best performance (using the information derived from the selected utility function);
– the optimal configuration is sent to all VoIP terminals (with WiFi beacons).

Such an architecture underlines two relevant aspects of the considered framework: 1) information captured by the metamodel and utility function is useful for supporting configuration / decision making processes in case of complex scenarios, such as the one considered in this paper; 2) cross-layering can be implemented in a *distributed* fashion (AP as reconfiguration manager for all the nodes).

5 Conclusions

This paper proposes a quantitative study of the problem of cross-layer performance optimization applied to a Voice over WiFi scenario, which enables to analyze and quantify inter-layer dependencies and to identify the optimal operating point of the system using cost-benefit principles. Achieved simulation results confirm some empirical considerations already available in the literature. The insight gained on the problem is then used to propose design principles for a Call Admission Control scheme able to enhance the overall system performance by limiting the number of users in the system and signalling to the active terminals the proper parameter settings to optimize overall performance.

Future work will deal with the actual implementation of the defined CAC scheme in order to provide performance analysis and validation of the proposed cross-layer optimization framework, as well as with the definition of proper signalling methods to support distributed cross-layering solutions.

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