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CO-CAC: A new approach to Call Admission Control for VoIP in 5G/WiFi UAV-based relay networks

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ABSTRACT

Voice over IP (VoIP) requires a Call Admission Control (CAC) mechanism in WiFi networks to preserve VoIP packet flows from excessive network delay or packet loss. Ideally, this mechanism should be integrated with the operational scenario, guarantee the quality of service of active calls, and maximize the number of concurrent calls. This paper presents a novel CAC scheme for VoIP in the context of a WiFi access network deployed with Unmanned Aerial Vehicles (UAVs) that relay to a backhaul 5G network. Our system, named Codec-Optimization CAC (CO-CAC), is integrated into each drone. It intercepts VoIP call control messages and decides on the admission of every new call based on a prediction of the WiFi network's congestion level and the minimum quality of service desired for VoIP calls. To maximize the number of concurrent calls, CO-CAC proactively optimizes the codec settings of active calls by exchanging signaling with VoIP users.

We have simulated CO-CAC in a 50 m × 50 m scenario with four UAVs providing VoIP service to up to 200 ground users with IEEE 802.11ac WiFi terminals. Our results show that without CAC, the number of calls that did not meet a minimum quality level during the simulation was 10% and 90%, for 50 and 200 users, respectively. However, when CO-CAC was in place, all calls achieved minimum quality for up to 90 users without rejecting any call. For 200 users, only 25% of call attempts were rejected by the admission control scheme. These results were narrowly worse when the ground users moved randomly in the scenario.

1. Introduction and related works

Unmanned Aerial Vehicles (UAVs) can be utilized to provide voice communications service to ground users, extending or replacing the traditional infrastructure of mobile networks, or in disaster scenarios [1–6]. Several authors have suggested the deployment of UAV-based networks with communication technologies that range from aerial base stations (BS) connected to a 3G backhaul [7] to more complex hierarchical architectures, including a WiFi access network, a distribution network, and a 5G backhaul [8]. In this work, we assume the deployment of a simple network such as the one shown in Fig. 1, made up of drones equipped with two network cards: a WiFi Access Point (AP) used for ground users coverage and a 5G link connected to a 5G backhaul network. Relay is performed between both networks for VoIP traffic and call control. WiFi is ubiquitous in today's end-user terminals (there were over 21 billion WiFi devices worldwide in 2021 according to [9]), while telco 5G infrastructure has expanded in recent years, counting with more than 500 million subscriptions in 2021 [10].

There are several papers in the literature devoted to optimizing the placement of UAVs in cellular deployments (4/5G), seeking traffic load balancing [11], maximizing throughput [12–14], reducing energy consumption [15,16], or maximizing coverage [17,18]. Other authors have addressed quality of service in 5G cellular networks, such as Tanveer et al. [19], who proposes an algorithm to dynamically reserve radio channels. However, a particular concern in the scenario shown in Fig. 1 is that the WiFi access network may turn into a bottleneck due to its shared medium access control (MAC) mechanism. Thus, if the VoIP traffic exceeded the capacity of the drone's AP, the quality of service would slump for all users covered by the congested Wireless Local Area Network (WLAN). This point of saturation is known as Voice over IP (VoIP) capacity of the WLAN, and its calculus requires to consider not only signal coverage, but also application-layer traffic and the MAC sub-layer of the WLAN [20]. In previous work, we addressed the problem of optimal initial placement of UAVs for the deployment of a multi-layer network considering the VoIP capacity of drones to provide VoIP over WiFi (VoWiFi) with the G.711 codec [8,21,22]. However, in this work, we focus on the problem of maintaining the speech quality for VoWiFi users *during* service operation in a scenario such as that described in Fig. 1. To the best of our knowledge, this has not been addressed before.

Keeping speech quality during calls is a classic problem in public switched telephone networks (PSTN). In its origins, the primary artifact used by telco operators to preserve the quality of service (QoS) was to

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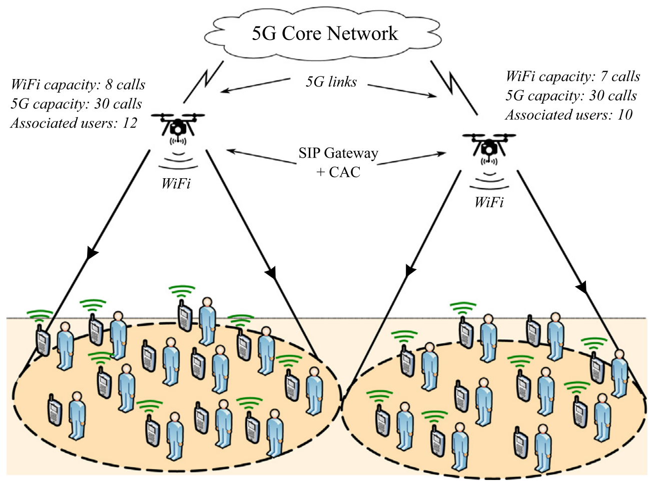

Fig. 1. Scenario of application.

reject new calls when all circuit-switched lines were busy. This function is known as Call Admission Control (CAC). 5G has built-in mechanisms that use multiple carrier schemes to offer high data rates and ensure high QoS [23]. But similarly to PSTN, the network performs admission control when radio resources are exhausted so requests for new resources are denied. Indeed, some admission control schemes have been suggested in the 5G literature for different purposes, such as user offloading between cellular and WLAN (such as WiFi) [24] to reduce interruption time during re-connection of users moving between macro and small cell domains [25], or for effective allocation of network resources for sliced 5G networks [26]. In this work, however, each drone acts as a single 5G user equipment, and we assume that the 5G backhaul is not congested. Therefore, an admission control scheme to maintain QoS for VoWiFi calls in our application scenario should be focused on WLAN rather than the 5G network.

Maintaining VoIP speech quality on a WLAN such as WiFi is a challenging task [27]. The standard IEEE 802.11e provides default mechanisms for QoS support, such as priority queuing [28]. But despite this, if VoIP traffic exceeds WLAN capacity [20], the WLAN performance falls steeply, affecting all VoWiFi calls that share the wireless link [29,30]. Thus, ultimately, a CAC mechanism is necessary to prevent VoWiFi calls that would overwhelm the link capacity. Indeed, the standard defines an Access Control mandatory mechanism to limit the number of concurrent VoIP flows and avoid saturation conditions, thus guaranteeing performance goals [31]. However, the actual decision algorithm is beyond the scope of the standard as it needs to be adapted to the application scenario. This is the goal of this work: the design of a call access control mechanism suited to the scenario described in Fig. 1. This endeavor entails the design of a CAC scheme that preserves QoS for VoIP in WLAN and its integration in the context of heterogeneous networks such as WLAN and 5G through VoIP call control messages.

Related works Several CAC algorithms have been defined for VoWiFi in different contexts than ours [27]. Measurement-based decision algorithms periodically monitor WLAN performance [32–34] while model-based algorithms predict WLAN performance by using analytical models of the IEEE 802.11 MAC sublayer [35–38]. Frequently, model-based CAC decision algorithms also include the optimization of other aspects such as operational MAC parameters [35,39], frame aggregation [36], or rate throttling [39] to increase the number of admitted calls. In some cases, however, the assumptions made by the authors, along with required device modifications, limit the applicability in real-life environments. In a previous work, we defined a centralized CAC mechanism in the context of a corporate data network connected to the PSTN where WiFi and Ethernet user terminals run a VoIP application with the G.729 audio codec [40].

Contributions This article proposes Codec Optimization and Call Admission Control (CO-CAC), a system for continuous assurance of VoIP speech quality in VoWiFi-to −5G UAV-aided relay networks. Its main characteristics are as follows:

- It uses a measurement-assisted analytical model that predicts the performance of the WLAN offered by each drone in terms of delay and loss. The expected performance is translated to a speech quality level for the active calls covered by the UAV.
- Proactive operation. The admission decision to the call is based on the speech quality level that would result if the new call was admitted. Before rejecting a call, CO-CAC proactively seeks bandwidth reduction by estimating alternative codec settings for active calls that would allow the new call to take place, enforcing the new configuration if plausible. This feature increases the number of accepted calls.
- 5G compatible. Our system includes a call handling module and the 5G compatible Adaptive Multi-Rate (AMR) wideband codec. CO-CAC is aware of VoIP control messages between WiFi devices and 5G terminals and generates or modifies them as needed.
- Its implementation does not require modifications to the VoIP terminals (i.e., transparency). Furthermore, it can be integrated into each drone that runs a VoIP Proxy/Gateway, so it does not require a dedicated device.

We use simulation to show the benefit of CO-CAC and a performance evaluation that shows its feasibility thanks to a short response time during VoIP call setup. We believe CO-CAC can be applied in circumstances where ground users have only WiFi terminals such as tables, or have cellular terminals that have WiFi, but the use of 5G is not desired by users or possible (e.g., roaming is disabled, lack of coverage at the ground level, 3/4G terminal, etc.).

The remainder of this paper is as follows. Section 2 elaborates on the application scenario described in Fig. 1 and provides an overview of the system architecture, call flows, and the system's operation. Section 3 describes CO-CAC and its modules. Section 4 shows the results obtained by simulation. In Section 5, we discuss these results and describe some of the main limitations of this work. Finally, Section 6 concludes the paper.

2. Scenario of application and system requirements

Traditional telco operators started adopting VoIP in the 90s, seeking cost reduction. Similarly, modern cellular networks have also been migrating to VoIP since UMTS (3G). In comparison with traditional voice calls, ensuring QoS in VoIP networks is challenging due to the best effort service of IP and the strict bounds in network delay, packet loss and jitter required by VoIP [41–43]. Thus, telco operators used dedicated bandwidth and traffic prioritization in their well-provisioned IP transport networks to guarantee an adequate QoS level. However, VoIP is not feasible out of the operator management domain (e.g., Internet or WLAN) if the network experiences persistent saturation conditions despite existing end-to-end QoS support mechanisms such as loss recovery [44], error concealment [45], congestion control [32], or bandwidth reduction [46–49]. Today's modern VoIP audio codecs bundle some of these mechanisms in adaptive multi-rate codecs, such as AMR [50], iLBC [51], OPUS [52], or iSAC [53], to name a few. VoIP codec frames are commonly transported as payload using the Real Time Protocol (RTP) and the User Datagram Protocol (UDP) over the IP network, constituting a VoIP media flow. Note, however, that the actual codec used on each VoIP call is negotiated in the call setup and could be altered during a session. This job is done by the Session Initiation Protocol (SIP) [54,55] which is prevalent today for VoIP call control. 5G networks support the AMR codec and SIP call control protocol. The authors assume that the reader is acquainted with VoIP technologies, particularly with SIP operation and messages [56].

As illustrated in Fig. 1, our scenario of reference is an outdoor area where users with WiFi devices run a VoIP app compatible with SIP signaling and the AMR codec. A set of UAVs equipped with 5G and WiFi access points is deployed as WiFi-5G gateways. Each UAV also acts as a SIP proxy/gateway for voice and call control messages. Our starting point assumes that drones are already placed, each covering a set of associated WiFi devices. This can be done with an optimal positioning problem such as in [21]. As mentioned above, the motivation of this work is to maintain VoIP speech quality during service operation by using Call Admission Control on each UAV, so new calls that could lead to unacceptable speech quality for VoWiFi users are rejected. Our system proactively seeks to maximize the number of concurrent VoIP flows by dynamically reconfiguring the settings of the AMR codec of active users.

Fig. 2 illustrates a conceptual call control message exchange between four entities in a basic application example. The Calling Party (associated to the drone's AP) calls a 5G user (Called Party). A third user in the same WLAN as the Calling Party is already engaged in a call. We describe each VoIP call control message produced for illustrative purposes rather than showing the actual SIP message. First, the Caller asks the SIP Gateway to start a new call, providing media information (e.g., codecs supported). Then, the SIP Gateway initiates the call setup through the 5G infrastructure, which routes the call control messages to the Called Party. The Called Party generates a call control message accepting the call, which goes back to the SIP Gateway. The reception of this message triggers the CO-CAC system, which estimates the speech quality level that active calls would experience if the new call was accepted. If the estimation results above a predefined minimum threshold, the call should be accepted or otherwise rejected. However, before rejection, CO-CAC tries to find alternative codec settings for active calls seeking bandwidth reduction in the WLAN. If an alternative configuration was found, the SIP Gateway would inform the affected parties (e.g., Third Party) by sending additional control messages to modify the codec settings (e.g., SIP Re-INVITE, see case B optional in Fig. 2). However, if no alternative configuration was possible, the call would be rejected (e.g., SIP CANCEL, see case A in Fig. 2).

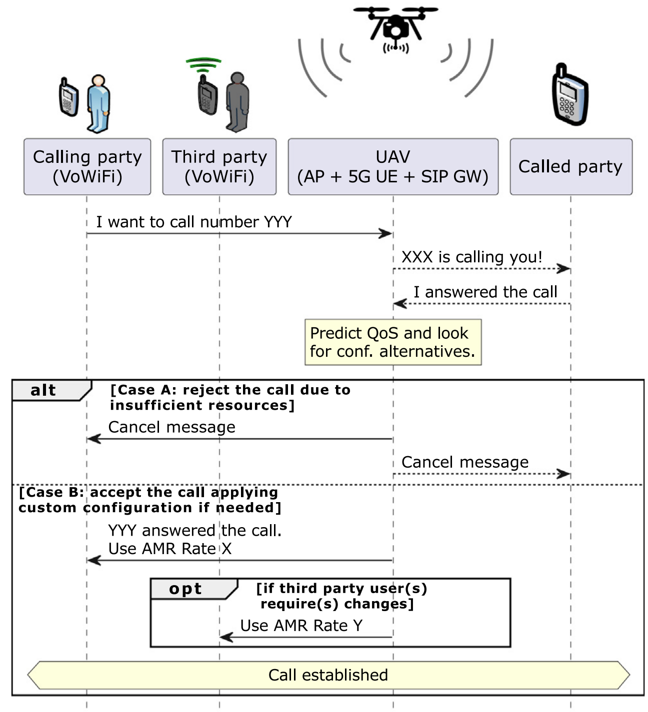

Fig. 2. Basic scenario of application.

2.1. System requirements

The CO-CAC system can be implemented in the SIP gateway running at each UAV. Popular open-source software for Private Branch eXchange (PBX) (e.g., Asterisk or FreePBX) already provides built-in methods for integrating external code. According to the previous example, our CO-CAC system has to be able to:

- Detect call events (i.e., SIP call control messages) to track call states and handle incoming calls.
- Collect measurements from the WiFi stations associated to the AP, such as SNR and the physical bit-rate or MAC operational parameters.
- Run a measurement-aided analytical model capable of predicting the speech quality level for active calls based on the network conditions. It also needs to simulate codec configuration alternatives if required.
- Modify and/or generate SIP control messages to force codec reconfiguration or to reject calls.

2.2. Network architecture and protocol stack

Fig. 3 illustrates the protocol stack for each functional entity involved in a call. The user terminal (WiFi device) uses a separate protocol stack for VoIP call control (SIP) and for VoIP media flow (AMR-WB codec frames). Both end in the UAV SIP Gateway.

The user and control planes in the 5G domain are shown as two different protocol stacks –top and bottom– respectively. In the user plane, the 5G network establishes and maintains session tunnels (e.g., User Plane Function, UPF) using the GRPS Tunneling Protocol (GTP) to transport VoIP flows or SIP call control messages. For example, the 5G network should route SIP messages to the IP Multimedia Subsystem (IMS). Thus, it has to establish and maintain a GTP-U tunnel from the gNodeB (gNB) to the corresponding UPF, which is finally able to deliver the traffic to the IMS Data Network (DN). Regarding voice frames, the 5G architecture features a new codec, namely EVS (Enhanced Voice Services), which is backward compatible with AMR-WB [57]. As such, transcoding is not needed in our scenario.

The control plane in 5G networks is responsible for establishing and maintaining the tunnels required by the user plane (i.e., from the User Equipment (UE) -at the rightmost part of the UAV's protocol stack- to the DN). This maintenance requires two Non-Access Stratum (NAS) protocols for mobility management (NAS-MM) and session management (NAS-SM), which results in two new functional entities: the Access and Mobility Management Function (AMF) and the Session Management Function (SMF), respectively. The underlying protocols provide specific functionalities, such as maintaining UE-to-gNB radio channels or sessions using the Radio Resource Control (RRC) protocol or the Packet Data Convergence Protocol (PDCP) protocol, respectively, to name a few.

3. CO-CAC system description

Fig. 4 shows the main components of CO-CAC as four software modules.

- The Call Handling module: is aware of incoming SIP messages and detects calls' state (1). When a new call is detected, it fetches information from calls in progress from the SIP database (2) and measurements related to QoS from every station associated with the Access Point from the AP database (3). Then, it asks the Decision module to evaluate speech quality using the previous information and to find the optimal AMR rate for every call (4). If the call is accepted, the Call Handling module updates the SIP database, registering the new call (5). Finally, it generates

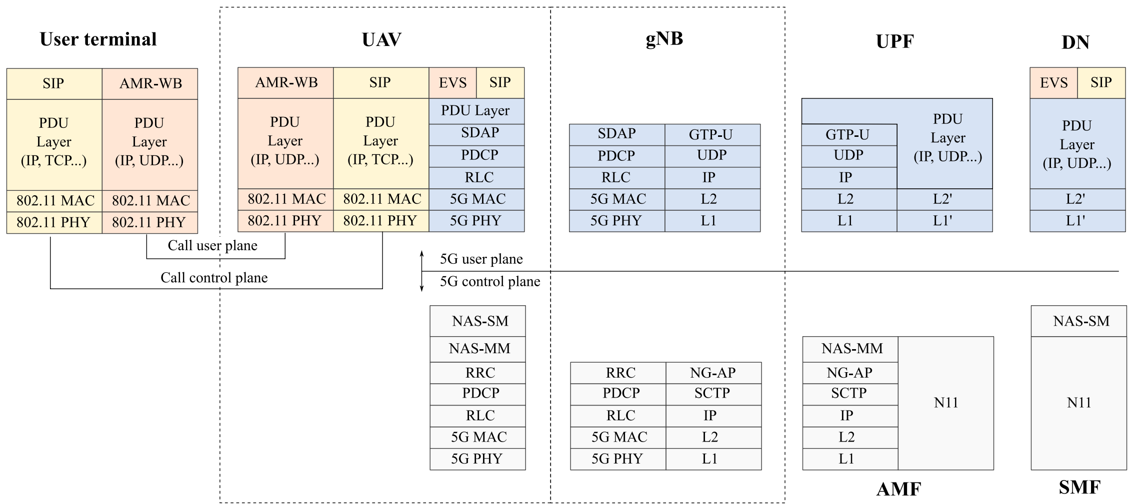

Fig. 3. Scenario protocol stack.

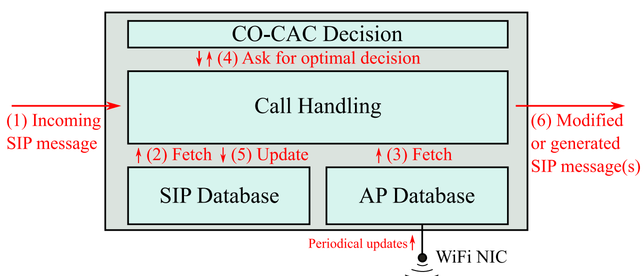

Fig. 4. CO-CAC system modules.

a modified or new SIP message to apply the admission decision (including reconfiguration, if needed).

- The SIP database: stores and maintains information about pending or established calls, such as call status (pending/in progress), peers involved (network addresses), and session media capabilities (list of available codecs).
- The AP database: contains updated information from the associated WiFi stations, such as the Signal-to-Noise Ratio (SNR), the physical data bit rate, and some MAC-layer parameters (e.g., Guard Interval).
- The Decision module: is responsible for evaluating the expected speech quality using the information received from the call handling module and making the final decision about acceptance, returning its result to the Call Handling module.

Since all modules reside in the same physical entity, there is no need to define custom protocols for communication between modules. Next, we elaborate on each module.

3.1. Call handling module

As mentioned above, this module is the orchestrator of the CO-CAC system. Through SIP control messages, it acts as a SIP proxy between user terminals and the 5G network.

After detecting a new call, it waits until the information about the media content is exchanged (this information is specified using the Session Description Protocol (SDP) enclosed in SIP messages). Then, it collects extra information from other modules and calls the decision module, which returns one of the following cases:

(a) If the call is accepted without changes, the call proceeds as normal. Then, it updates the call information in the SIP database.

(b) If accepting the call requires changes to active calls, custom SIP messages are generated and sent to the affected calls (SIP 183 Session Progress) to the calling party (with a custom AMR rate), which in turn notifies the chosen codec to the called party by using the SIP UPDATE control message. If needed, it also generates and sends SIP Re-INVITE messages to other calls in progress to force custom AMR modes. Finally, it updates the call information in the database.

(c) If the call cannot be accepted (no alternative configurations can guarantee a minimum speech quality level), SIP cancel messages are sent to both peers to announce the end of the call establishment.

Fig. 5 shows a signaling example showing the previous cases. Note that SIP messages modified or generated by CO-CAC are in blue, while the default flow of call control messages is black.

3.2. CO-CAC decision module

Algorithm 1 shows the pseudocode of the decision procedure. Its input parameters include the collection of calls in progress (Φ) along with their related information, the calling party information (A), the called party information (B), and the minimum quality threshold required to accept the call ($R_{\min}$). First, it builds a provisional candidate set (Φ') by joining any WLAN peers (A and/or B) to the collection of calls in progress (Φ). Second, the optimization loop occurs until the expected quality level of the candidate set ($WiFi\text{-}QoS(\Phi')$) is greater than $R_{\min}$ or every call is already optimized. Finally, the decision function returns its decision (CANCEL/ACCEPT), also providing the set of modified calls (Φ') if accepted.

VoWiFi QoS estimation VoWiFi speech quality assessment methods have been widely studied in the literature [42] using various techniques such as polls [58], PSQM [59], PAMS [60], P.562 PESQ [61] or the E model [62]. However, the E-model is the only method capable of estimating QoS for VoIP in the planning stage [63,64], as the rest require original and degraded voice samples.

The Wide-Band (WB) E-model [65] (since we use a wideband codec) proposes a speech quality score, namely the R_{wb} factor, which ranks from 0 (poor) to 129 (excellent). Then, R_{wb} can be translated to the universal R scale (from 0 to 100) using a linear approximation ($R = R_{wb}/1.29$) [65]. A well-known minimum value for the speech quality level to be acceptable is 60.

R_{wb} can be calculated by subtracting additive impairment factors from a maximum achievable score expressed in (1), where $R_{0,wb}$ represents the basic signal-to-noise ratio; $I_{e,eff,wb}$ can be expressed as

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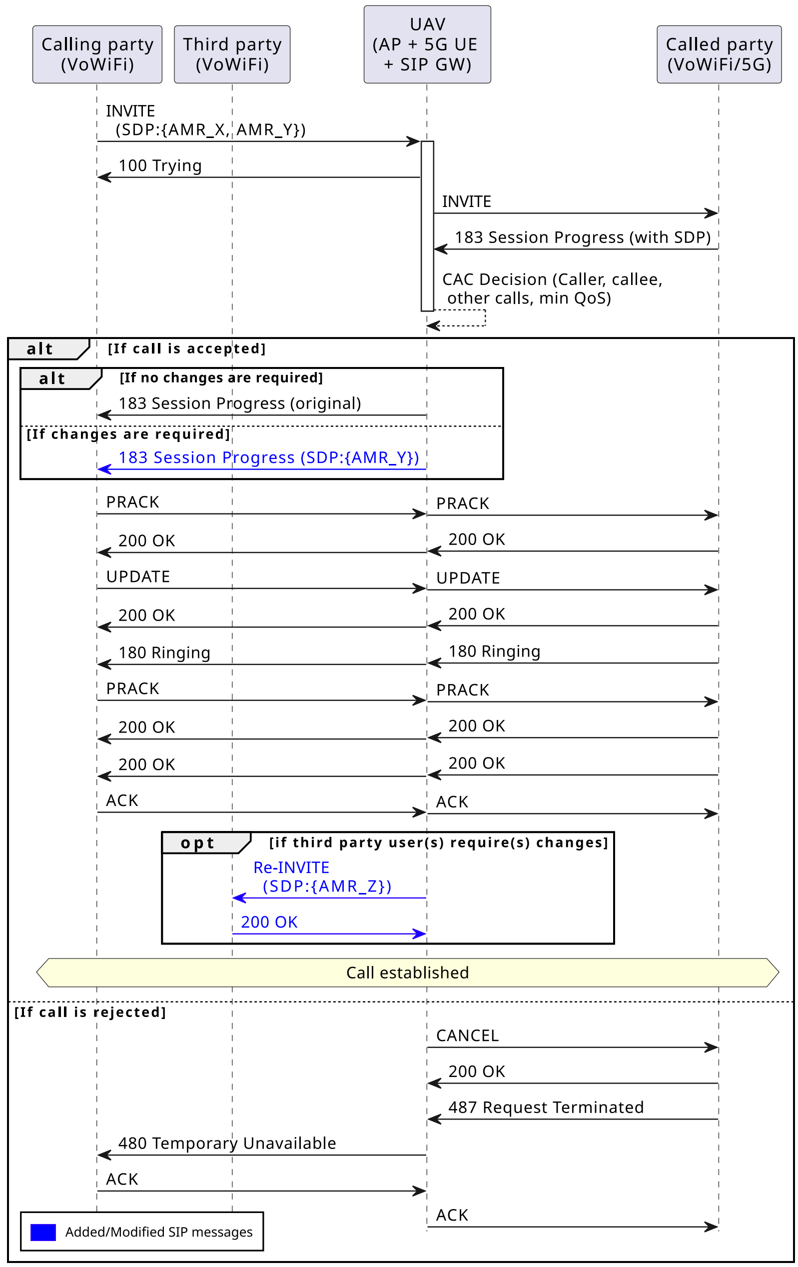


Fig. 5. Detailed call flow. (For interpretation of the references to color in this figure legend, the reader is referred to the web version of this article.)

a function of the packet loss rate (P_{pl}), the robustness of the codec packet (B_{pl}), and the degradation caused by codec compression (I_e); and finally $I_{d,wb}$ depends on the one-way delay (d).

$$R_{wb} = R_{0,wb} - I_{d,wb} - I_{e,eff,wb} \tag{1}$$

Although $R_{0,wb}$, B_{pl}, and I_e are codec-related parameters that can be assumed to be constant values [65], P_{pl} and d depend on network performance, which in our scenario is determined by the WLAN and the 5G backbone. Therefore, the packet loss ratio and delay can be calculated as follows.

$$P_{pl} = 1 - \left(\left(1 - L_{WiFi}\right) \cdot \left(1 - L_{5G}\right)\right) \tag{2}$$

$$d = d_{codec} + d_{WiFi} + d_{5G} \tag{3}$$

where d_{codec} represents the packetization interval (which is known for a given codec), and L_{WiFi} and d_{WiFi} represent the packet loss rate and delay in the WiFi access network; and L_{5G} and d_{5G} represent the worst-case metrics (upper bound) for the 5G link, which can be determined from the operator's Service Level Agreement.

In the previous expression, the terms related to 5G network performance and VoIP codecs are known. However, the WLAN performance depends on real-time network congestion and traffic load. Access to the shared medium is not deterministic in the WLAN but is ruled by the distributed coordination function (DCF) of the IEEE 802.11 MAC sublayer. Basically, each contending station must sense the medium during a fixed period to check that it is idle before transmission. If it is busy, the station waits for a random back-off interval before trying again. This back-off procedure is also activated between two consecutive transmissions.

WLAN performance modeling is a mature subject that has been addressed in the literature [20,27,31,66]. Most authors use Markov chain models of the MAC sublayer [67–72] to predict WLAN performance. These works provide an analytical expression of the probability that a station attempts to transmit during a randomly chosen time slot

Input: Φ (collection of calls in progress), A (calling party), B (called party), $R_{\min}$
Output: ACCEPT (Φ') / CANCEL

/* Initialize collection of optimized calls */
$\Phi' = \Phi$;
/* Add calling/called parties to the collection if they are associated to the UAV WiFi network (i.e., if they affect the WLAN performance) */
if *A is associated to the UAV WiFi network* **then**
  $\Phi' = \Phi' \cup A$;
if *B is associated to the UAV WiFi network* **then**
  $\Phi' = \Phi' \cup B$;
/* Evaluate speech quality at the WiFi network */
$R =$ *WiFi-QoS*(Φ');
/* If QoS is not satisfied by default, look for an alternative configuration */
while $R < R_{\min}$ **do**
  /* Randomly select an unoptimized call. Returns ∅ if all calls are already optimized */
  $x =$ *GetRandomUnoptimizedCall*(Φ');
  if $x \neq \varnothing$ **then**
    /* Reduce the AMR Rate of selected call, and estimate the new QoS */
    ReduceAMRRate($\Phi'(x)$);
    $R =$ *WiFi-QoS*(Φ');
end
/* Take final decision */
if $R \geq R_{\min}$ **then**
  return ACCEPT (Φ');
else
  return CANCEL;

Algorithm 1: CO-CAC Decision algorithm pseudocode

(frequently named τ). To fit the scenario in this paper, we use the expression τ from our previous work [40], which considers the following realistic assumptions: (a) Heterogeneous stations (each station can have a different physical data rate and/or codec), (b) non-saturated stations (i.e., buffers can be empty at any time), (c) noisy channels.

The probability τ allows one to derive closed-form expressions of the packet loss (L_{WiFi}) and delay (d_{WiFi}) in the WLAN from Eqs. (2) and (3) respectively. The algorithm 2 shows the process carried out to calculate the speech quality level (R) by solving the equation system developed in [40] in an iterative way (lines 1–8), then calculating the worst-case metrics (downlink—AP) on line 9, and finally estimating the level of speech quality.

The computational complexity of the *WiFi-QoS* function increases linearly with the number of stations (S), so its complexity is upper bounded by $O(S)$. Then, according to Algorithm 1, the *WiFi-QoS* function is executed up to $S \cdot C$ times at a worst case scenario (i.e., every call has to be optimized), where S represents the number of WLAN stations and C accounts for the number of AMR Rate Modes allowed at each VoWiFi device. Since the maximum number of AMR Rate Modes is constant (see Table 2), the CO-CAC algorithm computational complexity is limited by $O(S^2)$.

3.3. Databases and state extraction

Our system requires two core functionalities that support the CO-CAC Decision module:

- Explore and modify SDP content in SIP messages, further enabling the Call Handling module to be aware of call context and to trigger the CAC decision function.

Input: S (set of stations with SIP/AP characteristics)
Output: R (estimated QoS)

/* Solve the DCF analytical model (see [40]) */
1 $\tau^{(j)}_{\text{next}} = T^{(j)}_{c}, \ \forall j$;
2 **while** $\delta > \Delta$ **do**
3   $\tau^{(j)} = \tau^{(j)}_{\text{next}}, \ \forall j$;
4   **for** $j = 1$ **to** $|S|$ **do**
5   *solve* $\tau^{(j)}_{\text{next}}$;
6   **end**
7   $\delta = \max\left(|\tau^{(j)}_{\text{next}} - \tau^{(j)}|\right), \ \forall j$;
8 **end**
/* Estimate performance metrics */
9 *solve* $L^{(ap)}$ and $d^{(ap)}$; // As in Ref. [40]
10 $P_{pl} = 1 - \left(\left(1 - L^{(ap)}\right) \cdot \left(1 - L_{5G}\right)\right)$; // Eq. (2)
11 $d = d^{(ap)} + d_{codec} + d_{5G}$; // Eq. (3)
/* Estimate QoS (R factor) */
12 $R_{wb} = R_{0,wb} - I_{e,eff,wb}\left(B_{pl}, I_e, P_{pl}\right) - I_{d,wb}\left(d\right)$;
13 $R = R_{wb}/1.29$
14 **return** R;

Algorithm 2: WiFi-QoS calculus

- Store user's WiFi device capabilities (AP Database) and track all calls in progress (SIP Database), which supports the stateful behavior of the CO-CAC.

3.3.1. SDP content exploration and customization

During the start of a SIP session, peers negotiate the codec. To do so, user agents typically embed their codec capabilities into the body of the SIP message as SDP content (i.e., Content-Type: application/sdp). SDP provides a plain-text structure that can describe session capabilities that is used by SIP. In SDP syntax, each line represents a normalized key–value parameter. CO-CAC uses the attribute parameter, which allows peers to specify attributes for each supported codec, for example, setting the supported `mode-set` (rate modes) in adaptive codecs.

3.3.2. State extraction

The CO-CAC system relies on two databases/tables: one for storing SIP sessions (SIP DB) and another for tracking WiFi devices associated to the AP (AP DB). A SIP session can be fully stated with the following fields:

- An identifier for each call (`call-id`), which can be generated by the call handling module, capable of providing a unique key in the SIP DB.
- A `call-type` field that stores the direction of the call: (a) an incoming 5G to WiFi call, (b) an outgoing WiFi to 5G call, and (c) a local WiFi-to-WiFi call.
- A `call-status` field to track which sessions are still pending.
- The AMR-WB mode (`call-mode`) chosen by the CO-CAC Decision module.
- Identifiers for the caller (`caller-id`) and called (`called-id`) (e.g., IP addresses).
- The list of AMR-WB modes available in the party calling (`caller-mode-set`) and called (`called-mode-set`) party, which will be used by the CO-CAC Decision module.

On the other hand, each WiFi station is assumed to be monitored by the associated Access Point installed on the drone. Tracking information includes `user-id`, which is related to the previously mentioned `caller-id` and `called-id`; the station IP address (`ip-addr`); the station MAC address (`mac-addr`); and physical layer attributes such as SNR (`phy-snr`) and bit rate (`phy-bitrate`).

Table 1
Simulation parameters.

	Parameter	Value
Scenario	Scenario size	2500 m^2
	Ground users	0–200
	UAVs	4
QoS	R_{min}	65
Traffic model	λ	5 calls/h
	T	180 s
	AMR codec modes	0–7
Mobility pattern	Move prob.	0.6
	Change direction prob.	0.4
	Speed	5.4 km/h

Table 2
AMR-WB codec modes.

Mode	Codec	Rate (kb/s)
0	`AMR-WB_6.60`	6.60
1	`AMR-WB_8.85`	8.85
2	`AMR-WB_12.65`	12.65
3	`AMR-WB_14.25`	14.25
4	`AMR-WB_15.85`	15.85
5	`AMR-WB_18.25`	18.25
6	`AMR-WB_19.85`	19.85
7	`AMR-WB_23.05`	23.05

4. Results

In this section, we evaluate the performance of CO-CAC through simulation. We also scrutinize its behavior, showing that it can guarantee the required minimum speech quality by rejecting incoming calls when WLAN resources are exhausted, modifying calls in progress if necessary.

4.1. Simulation setup

We have created a simple discrete event simulator using Matlab©9.10 (R2021a). Our simulation tool implements all the modules of CO-CAC. The scenario simulated is a squared 2500 m^2 terrain, which is further divided into four subareas of equal size, each containing a UAV-mounted AP located on its centroid. At the beginning of each simulation, users are randomly placed, and each user is associated with the closest AP. The simulation time is 15 min. Each simulation has been run 30 times with different random seeds to generate a proper confidence interval.

The parameters and configuration used in the simulations can be summarized as follows (see Table 1).

- User behavior: Each user makes five calls per hour. Each VoIP call has a duration of 180 s (as suggested in the ITU-T P.561 recommendation) and is triggered using a random uniform function. Regarding user's movement, two cases will be considered: (a) static, where users do not move during simulation, and (b) dynamic, where users move randomly at a constant speed (5.4 km/h) along the terrain but stay still while engaged in a call. In the latter case, the closest UAV-mounted AP will be associated with the user every time a new call is started.
- Minimum speech quality level: The minimum level of speech quality required for calls is $R_{min} = 65$. CO-CAC will reject new calls that would lead to a quality level below this minimum threshold. Note, however, that without call admission control, this circumstance could occur at any time during the simulation. In such a case, the affected call(s) are automatically canceled, assuming that ground users hang up due to poor quality.
- Wifi device settings: we assume every device runs the IEEE 802.11ac standard revision at the 5 GHz band configured with 80 MHz channels. VoWiFi terminals also implement the AMR-WB modes listed in Table 2.

The following merit figure (call success) will be used to show the effectiveness of CO-CAC:

$$Success = 100 \cdot \frac{Accepted - Degraded}{Offered} \tag{4}$$

where *Accepted* stands for the number of calls accepted by the CO-CAC, *Degraded* refers to calls canceled due to falling below the threshold R_{min}, and *Offered* is the total number of calls made by ground users during the entire simulation. These indicators will be collected on each drone during the simulations. The results presented in this Section aggregate these indicators in the four drones.

To prove the feasibility of implementing our call admission scheme, we also measure the response time (i.e., the time since a call is initiated until the CO-CAC system produces a response) during the execution of the simulations. This delay should be added to the call setup procedure and should be as low as possible.

4.2. Numerical results

Fig. 6(a) shows the percentage of call success (i.e., calls not aborted or rejected) as the number of users increases from 0 to 200. We show four data series for this figure of merit: with and without CO-CAC, and with and without moving users. The colored shade surrounding the plot lines represents the 95% confidence interval. When CO-CAC is in place, the success ratio has been calculated using the number of rejected calls instead of degraded calls. The results show that CO-CAC is able to almost duplicate the number of users that experience a 100% of success ratio with respect to lack of admission control. A distinctive characteristic of CO-CAC is that it tries bandwidth reduction in active calls before rejecting new calls. Fig. 6(b) shows the percentage of calls that have experienced a change in the codec settings during operation. Logically, this percentage increases with the number of users. Again, the colored shade surrounding the plot lines represents the 95% confidence interval.

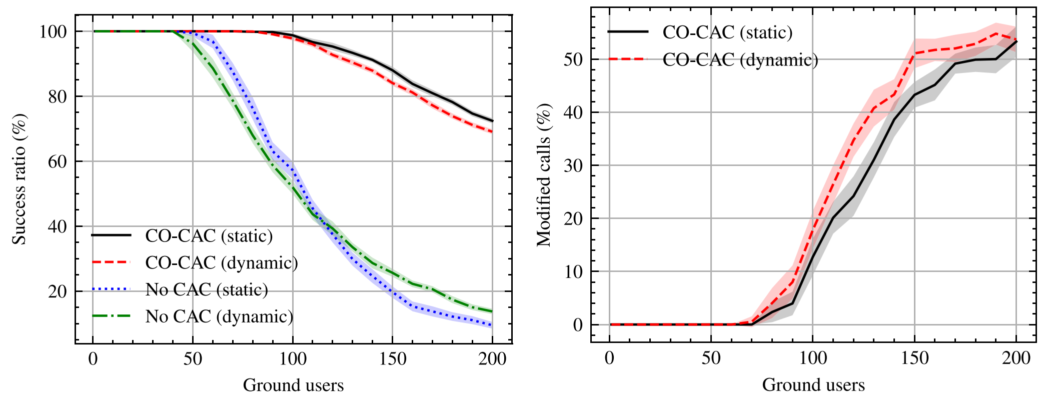
Fig. 7(a) has illustrative purposes and shows the temporal evolution of some statistics collected during a simulation run of 150 static users and CO-CAC. It shows the number of accepted, rejected, active, and degraded calls (which does not apply when admission control is in place).

Finally, Fig. 7(b) shows the average delay that CO-CAC would add to the call setup process due to computation delay (i.e., response time). The simulations were executed on a regular PC with an Intel 2.4 GHz processor and 4 GB of RAM. Note that this delay would not affect the VoIP media flow but only the initial call setup. The worst-case call setup delay was about 150 ms, which is acceptable and would likely pass unadvertised to end users.

5. Discussion

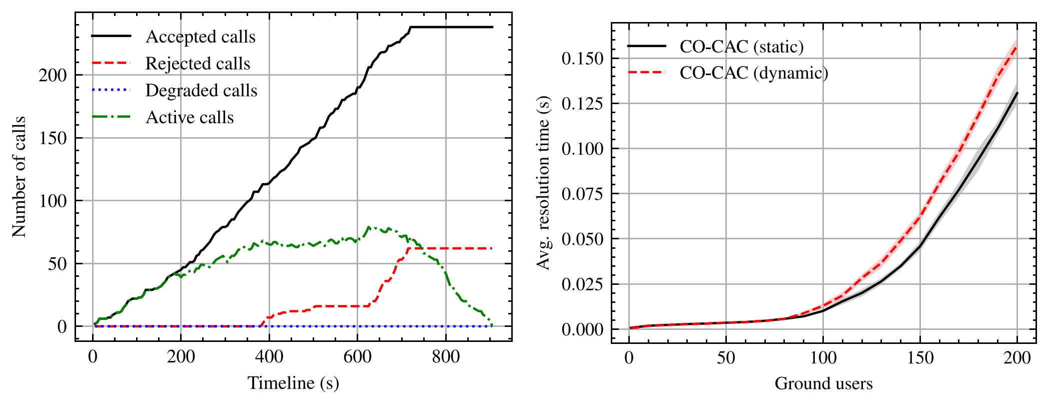
Looking at Fig. 6(a), we can identify three behaviors according to the range of users included in the simulation: 0–50 users, 50–100 users, 100–200 users. In light of these ranges, the percentage of successful calls from Fig. 6(a) can be interpreted as follows.

- Between 0 and 50 users: the network is not congested, so there is little difference between using CO-CAC or not using CO-CAC since all calls are successful.
- Between 50 and 100 users: some drones' WLAN enter saturation conditions. Thus, degraded calls increase with the number of users up to some 50%–60% (for 100 users) in schemes without CAC. In this range, however, CO-CAC is able to maintain almost 100% of call success thanks to its ability to reconfigure active calls to a lower mode of operation. Indeed, Fig. 6(b) shows that, for 100 users, almost 20% of the calls were modified which allowed that nearly 100% of calls were successful.



(a) Percentage of succeeded calls (b) Percentage of modified calls

Fig. 6. Success ratio (left) and percentage of modified calls (right) obtained for a variable number of VoIP users. (For interpretation of the references to color in this figure legend, the reader is referred to the web version of this article.)



(a) Timeline of CO-CAC experiment (150 static users) (b) Resolution time

Fig. 7. Evolution of some indicators during the simulation run (a), and average response time of CO-CAC during simulations (b).

- Between 100 and 200 users: reconfiguration of active calls does not suffice to accommodate new calls without impairing the speech quality below the threshold $R < 65$. Therefore, CO-CAC starts rejecting new calls, which produces a steady decline in the call success rate with the number of users. In the worst-case (200 users), almost 30% of calls were rejected by our admission control scheme (and more than 50% of calls were reconfigured). Still, less than 20% of the calls would have been successful without CAC.

The linear decrement in the success rate of CO-CAC can be better analyzed by examining the simulation timeline in Fig. 7(a) (150 users). Until about 375 s, all calls were admitted by CO-CAC thanks to bandwidth reduction, which affected almost 50% of the calls. However, some calls were rejected after that instant to preserve the minimum speech quality level. In particular, the maximum number of concurrent calls in this experiment was 70. Note that in a similar case (150 users) without CAC, the success ratio would be about 25% which would yield roughly 50 concurrent calls. Thus, CO-CAC increases the maximum number of simultaneous calls happening with acceptable speech quality.

The results from our simulations suggest that dynamic scenarios (i.e., user movement) perform slightly worse than static ones. This can be traced back to the fact that some users could move to a loaded AP, speeding up WLAN saturation in some simulations. This effect could be even worse with mobility schemes where users engaged on a call still move rather than staying still.

The parameters chosen for the simulations clearly influence the results obtained. For example, increasing the minimum speech quality ($R_{\min}$), or suppressing the lowest modes of the AMR-WB codec (see Table 2) would speed up congestion, so the three behaviors previously described would still hold, but would happen with fewer users. A similar effect would be obtained with a greater call arrival rate (λ) as it would increase the number of offered calls.

5.1 *Limitations*

Regarding applying the described system and the results shown in this paper, some limitations should be considered. First, although feasible in practice, the direct application of CO-CAC is constrained by several assumptions made in this work. For example, user terminals must run a VoIP app that implements SIP signaling and AMR codec. Although SIP has become pervasive in VoIP, some terminals (specially non-5G compliant) might implement different codecs that would require the adaptation of our system (e.g., quality model or reconfiguration of active calls). Second, besides the parameters influence mentioned above, the results shown strongly depend on the scenario chosen for simulation. Thus, a different model for user mobility, terrain size, or the number or position of drones would likely produce different results. In this sense, the results shown serve more as an illustrative purpose than a thorough performance evaluation presentation.

5.2 *Further work*

Some extensions of this work are possible and could be addressed in further work. An immediate continuation is to adapt the QoS model to other real-time services such as video. Another research line would be integrating the cellular network and WLAN in a single CAC scheme, which could also be useful for user offloading, as suggested in [24]. Another interesting timely topic for further research is the integration of cognitive capabilities in the 5G network [73–75] and/or in the WLAN [76]. Finally, another research line would be to explore the applicability of our proposal in multi-hop UAV relay networks [77], which would require to consider new performance metrics.

6. Conclusions

This work has presented a novel approach to call admission control in VoIP suited to scenarios where WiFi users' terminals are covered by UAV-mounted APs that relay to a 5G network. We have devised a system that leverages the AMR codec multi-rate to increase the number of concurrent calls while assuring a minimum speech quality during VoIP calls. Our CO-CAC system includes the admission control algorithm and the integration with the application scenario through standard SIP call control messages.

Overall, our results indicate that our system significantly increases the number of concurrent calls held with acceptable quality. Thus, CO-CAC could admit and guarantee minimum quality to the VoWiFi calls made by 90 users, while lack of CAC would have only been able to serve 50 users with acceptable speech quality. To achieve this, CO-CAC dynamically modified the settings of up to 50% of active calls. Our results also show the feasibility of implementing CO-CAC as a software component in existing VoIP gateways, having a minor impact on call set-up delay. This work could be extended in numerous ways, such as integrating the radio resources of the 5G network into the CAC scheme, which could be enhanced with cognitive capabilities.

CRediT authorship contribution statement

Vicente Mayor: Conceptualization, Investigation, Methodology, Software, Visualization, Writing – original draft. **Rafael Estepa:** Conceptualization, Investigation, Methodology, Supervision, Writing. **Antonio Estepa:** Supervision, Conceptualization, Validation, Writing – review & editing.

Declaration of competing interest

The authors declare that they have no known competing financial interests or personal relationships that could have appeared to influence the work reported in this paper.

Data availability

No data was used for the research described in the article.

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