A CALL ADMISSION CONTROL FRAMEWORK FOR VOICE OVER WLANS

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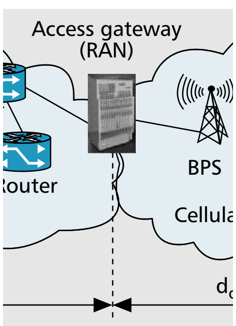

The call admission control framework presented in this article, called WLAN Voice Manager, manages admission control for voice over IP (VoIP) calls with WLANs as the access networks.

ABSTRACT

In this article a call admission control framework is presented for voice over wireless local area networks (WLANs). The framework, called WLAN Voice Manager, manages admission control for voice over IP (VoIP) calls with WLANs as the access networks. WLAN Voice Manager interacts with WLAN medium access control (MAC) layer protocols, soft-switches (VoIP call agents), routers, and other network devices to perform end-to-end (ETE) quality of service (QoS) provisioning and control for VoIP calls originated from WLANs. By implementing the proposed WLAN Voice Manager in the WLAN access network, a two-level ETE VoIP QoS control mechanism can be achieved: level 1 QoS for voice traffic over WLAN medium access and level 2 QoS for ETE VoIP services in the networks with WLANs as the local access. The implementation challenges of this framework are discussed for both level 1 and level 2. Possible solutions to the implementation issues are proposed and other remaining open issues are also addressed.

INTRODUCTION

Over the past decade, technologies have been developed to allow the transmission of real-time communications such as voice over IP (VoIP) networks or the Internet, which used to carry only data traffic. At the same time, the market for IEEE 802.11 WLANs has been experiencing a tremendous growth in recent years, as evidenced by the rapidly increasing popularity of 802.11 WLAN hotspots deployed in residence buildings and enterprises, as well as public areas. One key capability in next-generation wireless networks is to carry VoIP over WLANs and to realize seamless wireless data and voice communications. In fact, the technology to enable one phone number for both broadband wireless data and voice communication is already available now.

The ultimate objective of VoIP is to deliver reliable, high-quality voice service, which is comparable to what is provided in traditional circuit-switching networks. However, due to the limitation on bandwidth and the deficiency in ETE quality-control mechanisms in existing public IP networks and WLAN networks, the delivery of VoIP often results in unpredictable delay/jitter and packet-loss [1] performances. This is because both the Internet and the WLANs were originally designed mainly for data communications. It is therefore important to introduce ETE QoS provisioning and control methodologies for VoIP WLAN applications.

In recent years, substantial research and development efforts have been focused on VoIP QoS [1–4], and QoS for WLANs [5–7], which are all important to provide ETE QoS for VoIP over WLANs. In [2], a QoS provisioning and control mechanism was proposed for VoIP applications in soft-switch-based IP networks. In [3] a survey was presented on VoIP QoS management mechanisms. Admission control plays a critical role in delivering QoS for VoIP. As indicated in [4], in order to make VoIP attractive to end users, the only feasible and directly implementable alternative is to deploy an efficient mechanism within the endpoints. The authors in [4] proposed a call-quality monitoring and control framework for maintaining voice quality at acceptable levels over networks that do not offer QoS guarantees. However, all the schemes in [2–4] did not consider the cases when the VoIP calls are originated from WLANs, in which the MAC protocol, mobility, and handoff problems will jointly affect the QoS of the VoIP calls. In [5], a call admission and rate control scheme was proposed to support stringent QoS requirements of real-time and streaming traffic over IEEE 802.11 WLANs. The call admission control algorithm was used to regulate the admission of real-time or streaming traffic and the rate control algorithm was used to control the transmission rate of best-effort traffic over WLANs. Two critical problems in VoIP over WLANs were investigated in [6], that is, low VoIP capacity in a WLAN and unacceptable VoIP performance in the presence of coexisting traffic from other applications. The authors in [6] proposed some

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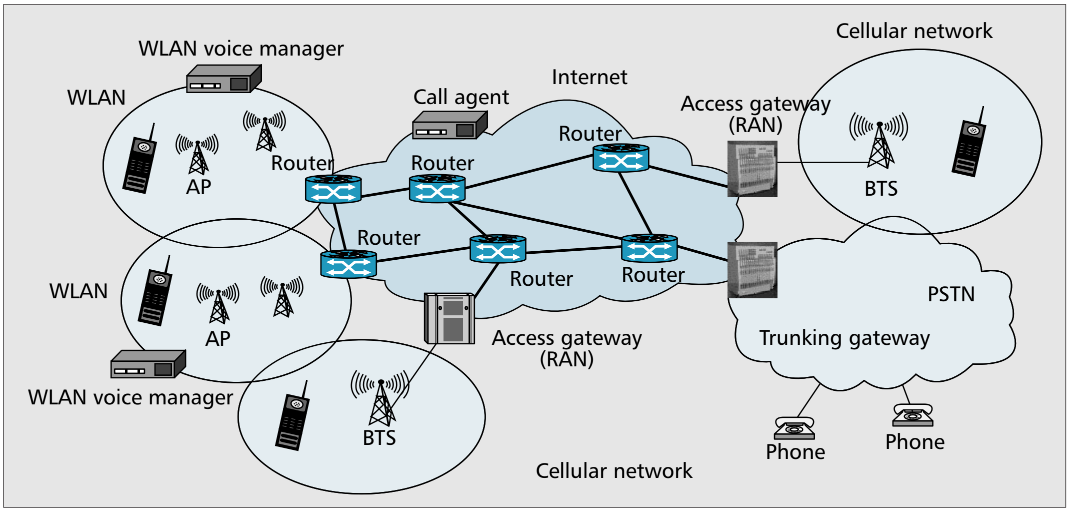

Figure 1. *An example of VoIP network with WLAN as the access network.*

new schemes to improve the two problems mentioned above. In [7] the authors proposed a polling with nonpreemptive priority-based medium access control scheme to provide predictable QoS in IEEE 802.11 WLANs. The proposed transmit-permission policy and adaptive bandwidth allocation scheme derived sufficient conditions such that all the time-bounded traffic sources satisfy their time constraints in order to provide various QoS guarantees in the contention-free period, while maintaining efficient bandwidth utilization at the same time. The proposed scheme is provably optimal for voice traffic in that it gives minimum average waiting time for voice packets.

With all the research efforts described above in QoS for VoIP and QoS for WLANs, it is clear that an ETE QoS provisioning mechanism is necessary to provide satisfactory VoIP services in the networks where WLANs act as the access networks and the IP networks are in the core. Traditionally for a typical existing commercial IP network or Internet that provides VoIP services, soft-switches or call agents coordinate with access gateways/trunking gateways to perform call setup, call teardown, and other signaling for the VoIP calls without an ETE QoS control mechanism involved. In this article, an ETE QoS provisioning and controlling framework is proposed for VoIP applications initiated from WLAN access networks. The core of this QoS framework is the WLAN Voice Manager for WLAN access networks, which interacts with soft-switches, gateways, and routers in the Internet to provision ETE VoIP QoS. The WLAN Voice Manager consists of five components: admission controller, admission policy, WLAN monitor, bandwidth broker, and a standard call agent function that performs call setup, call teardown, and other signaling for the VoIP calls. Its corresponding QoS control can be realized in two different levels. Level 1 interacts with the WLAN local MAC protocols, and handles admission control on the local level to WLAN originated VoIP calls. Level 2 handles dynamic intra/intersubnet bandwidth reallocation and coordinates ETE call admission control. With this two-level mechanism, VoIP QoS can be provisioned on an end-to-end basis.

The rest of the article is organized as follows. A VoIP network reference model with WLAN access networks is illustrated in the next section, where the functions of each network element are briefly described. Based on the VoIP network reference model, the WLAN Voice Manager is introduced, in which the interactions among different components of the WLAN Voice Manager are described, along with the added QoS control interfaces. Discussions are given on the implementation challenges for the framework and some possible solutions are proposed. The final section draws the conclusions of this article.

A Voice over WLAN Network Reference Model

VoIP is one of the fastest growing Internet applications today. At the same time, driven by huge demands for portable access, the WLAN market has been taking off quickly. Due to its convenience, mobility, and high-speed access, the WLAN represents an important future trend for the "last-mile" Internet access. In this article, a typical VoIP over WLAN network reference model is introduced, as illustrated in Fig. 1. In the public IP network, the call agent has the interfaces to access gateways/trunking gateways. Access gateways connect various access networks (cellular, cable, xDSL, POTS, etc.) to the IP networks. Trunking gateways connect the IP networks to the traditional PSTNs. By working with these different gateways, the call agent provides the VoIP users regular soft-switch services, such as user registration, voice call setup, call tear down, and so forth. WLANs are attached to the public Internet as the popular access networks. VoIP calls originated from WLANs will be the fundamental traffic source with ETE QoS problems concerned in this article.

For the VoIP over WLAN network reference model described in Fig. 1, an ETE QoS provisioning mechanism is needed since both WLANs and the Internet (IP networks) were originally designed for data communications, which is not

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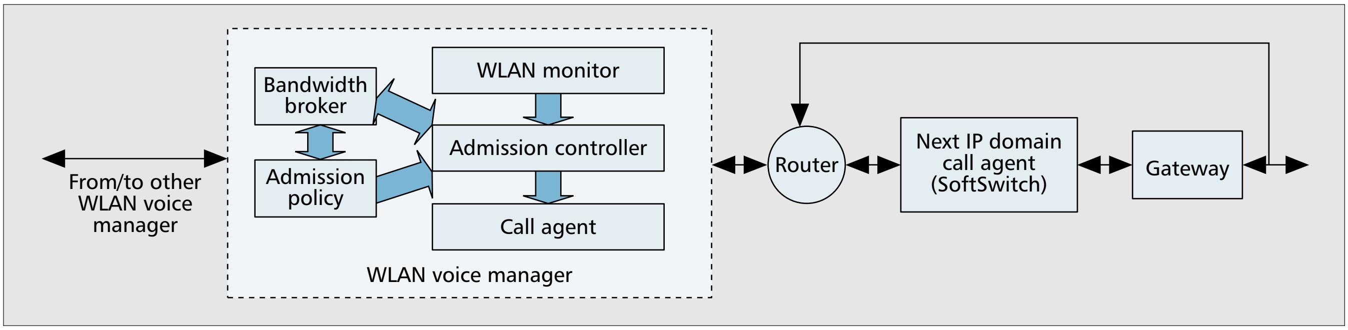

Figure 2. *The proposed WLAN voice manager functional diagram.*

well suited for real-time communications such as voice. QoS provisioning mechanisms are needed to augment call agents/gateways/routers in order to assure predictable ETE performance for the delay-sensitive and loss-sensitive VoIP applications. Some studies have shown that the ETE delay is the primary QoS concern in the Internet [3]. ITU G.114 defines the maximum tolerable ETE delay for real-time voice as 150 ms. The existing Internet and WLANs have difficulties in satisfying the QoS requirements of VoIP, due primarily to the network congestion caused by insufficient network bandwidth. Many mechanisms, including resource provisioning, traffic engineering, admission control, resource reservation, and connection management, have been proposed to resolve this problem. These mechanisms offer the solution of either predicting the traffic and engineering the network to ensure committed QoS, or restricting the total amount of traffic competing for the same resource. The next section describes the proposed framework, called WLAN Voice Manager, which is extended from the existing standard call agent functions and fits well into the existing VoIP network elements to provision and control ETE VoIP QoS.

Framework for WLAN Voice Manager

WLAN Voice Manager Components

The WLAN Voice Manager communicates with its own MAC layer medium access process and the WLAN Voice Managers in the neighboring subnets. It also communicates with the call agents, gateways, and routers in other IP subnets on its ETE path, in order to provision ETE QoS for the voice calls originated from the WLANs. These interactions coordinate the overall call management and bandwidth provisioning. The proposed WLAN Voice Manager consists of the following five components: admission controller, admission policy, WLAN monitor, bandwidth broker, and a standard call agent function. Figure 2 depicts the interactions among the different components inside a WLAN Voice Manager, and also the interactions between a WLAN Voice Manager and other VoIP network elements.

In a traditional telephony network the users access the system via call admission control. However, most of the current IP networks have no admission control and can only offer best-effect service. In other words, new traffic may keep on entering a network even beyond the network capacity limitation; consequently, both the existing and new flows suffer packet delay and losses. To prevent these, admission control mechanisms need to be in place. The proposed WLAN Voice Manager is a logical extension of the standard VoIP call agent function, which provides VoIP users regular soft-switch services, such as user registration, voice call setup, and call teardown. At the center of the WLAN Voice Manager is the admission controller, which takes inputs from WLAN monitor, admission policy, and bandwidth broker to make decisions on whether to accept or reject a VoIP call originated from the WLANs.

The WLAN monitor measures real-time QoS related performance parameters over the WLANs in the MAC layer (e.g., channel busyness ratio, etc. [5]). The admission policy stores a set of "rules" or "policies," based on which call admission control decisions are made and dynamic bandwidth allocations are negotiated. In the admission policy component, the QoS-related provisioning rules are emphasized. The bandwidth broker is involved in intrasubnet bandwidth provisioning. To achieve ETE VoIP QoS throughout all the IP network domains on the call path, bandwidth provisioning across IP subnet boundaries is critical. A bandwidth broker is one of the mechanisms for implementing bilateral service level agreement (SLA) negotiations between neighboring subnets, allocating network resources (bandwidth) on demand, and coordinating the ETE call admission control.

With the help of the above five-component WLAN Voice Manager, a two-level QoS mechanism can be achieved for VoIP over WLANs:

- Level 1: Real-time QoS provisioning through real-time monitoring on WLAN medium access processes, and enforcing call admission control on WLANs
- Level 2: ETE QoS guarantee by ETE dynamic bandwidth allocation and call admission control

The level 1 WLAN Voice Manager performs measurement-based admission control; while the level 2 WLAN Voice Manager performs parameter-based admission control. The proposed WLAN Voice Manager aims to provide QoS for voice over IP traffic in two domains, level 1 in the WLAN local access domain, and level 2 in the ETE network domain, which includes the

WLAN as the access network. The two levels can be implemented separately, and both levels need to be in place in order to provide ETE QoS for the voice applications over the IP network with a WLAN access network.

Level 1 WLAN Voice Manager

Assume that a WLAN operates in the infrastructure mode, that is, there is a fixed entity called an access point (AP) that bridges all data traffic between the mobile stations associated with it. An AP and associated mobile stations form a basic service set (BSS) communicating in the unlicensed RF spectrum. A collection of APs connected through a distribution system (DS) can extend a BSS into an extended service set (ESS), as shown in Fig. 3. A handoff occurs when a mobile station moves beyond the radio range of one AP, and enters another BSS at the MAC layer.

The level 1 WLAN Voice Manager performs real-time monitoring and admission control for the WLANs by involvement with the WLAN monitor. By measuring and monitoring the WLAN MAC performance on medium access, and doing local admission control based on real-time monitoring and the admission policy, the level 1 WLAN Voice Manager can proactively manage the network resources and provision VoIP QoS to prevent potential QoS degradations. The following measurement-based level 1 call admission control procedure is proposed. For every fixed measurement period, the WLAN monitor collects measurements, as mentioned above. The admission controller is updated with the most recent measurements from the WLAN monitor. Whenever a new VoIP call initiates from the WLAN, the admission controller compares the measured real-time network conditions on MAC process with policies from the admission policy and decides whether to accept or reject the call request based on these policies. The admission controller then informs the call agent and the VoIP access gateway and the router of the level 1 admission controller decisions, accordingly. The call agent will set up the call if the new VoIP call request is accepted in the WLAN.

Level 2 WLAN Voice Manager

The level 2 WLAN Voice Manager handles dynamic bandwidth allocation and coordinates call admission control on an end-to-end basis. The other component, the bandwidth broker, is involved at this level. Suppose that in each IP subnet (ISP, carrier, enterprise, etc.) there is a bandwidth broker acting as the QoS gateway. This QoS gateway serves to provide a signaling mechanism for carrying ETE call admission control decisions. It is also involved in allocating network resources on demand. It can negotiate with the neighboring QoS gateways to adjust resource commitments at the boundary routers as well as to reconfigure internal subnet elements, all based on the real-time information measured from the IP subnet and dynamic SLAs. RSVP was suggested in [8] as the major signaling protocol for the level 2 WLAN Voice Manager. The bandwidth broker uses two admission control algorithms: per-flow ETE guaranteed delay services and class-based admission control in a core stateless network, as proposed in [9]. The bandwidth broker is used to store and manage QoS reservation state, and execute admission control and resource management. Upon a new call request arrival, admission control unit first computes the equivalent bandwidth requirements of the new call, and then compares with available resource to make a decision.

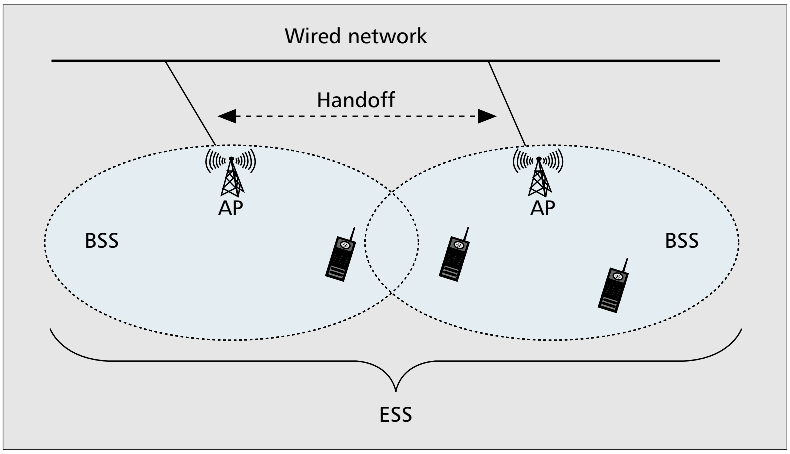

Figure 3. *A typical IEEE 802.11 WLAN.*

Technical Challenges and Some Possible Solutions

Challenges for WLAN Voice Manager Implementations

The proposed WLAN Voice Manager framework faces several challenges. For level 1 voice management in the WLANs, the key issue will be the coordination of call admission control in the WLANs for voice calls in terms of wireless medium access. Multiple call admission control policies can be stored in the admission policy component, interoperate within a single WLAN domain, and share the same policy information. The central repository, the admission policy, can be used to store, distribute, and coordinate policy information among such systems. How to set up the rules or policies of voice call admission control and QoS provisioning is not a trivial task. For level 2 voice management in WLANs, the bandwidth broker and admission controller will be involved. There are several implementation issues here. One of the most challenging issues is the signaling procedure for the bandwidth broker in order to implement real-time ETE QoS control. For the admission controller, the ETE call admission control algorithm is the key point. In the following section, some of the implementation issues will be addressed in detail.

Technical Solutions

Level 1 Solutions — Consider the case of a VoIP call originated from a WLAN (either a new call request or a handoff call from a neighboring AP); the level 1 Voice Manager can be implemented using the following call admission control (CAC) algorithm, CAC_{L1}, which was motivated and revised from [5].

A longer measurement period cannot reflect the network dynamics well, but costs less network resources. A good trade-off can usually be determined based on the simulation analysis. An accurate measurement of the network is also very challenging due to the network dynamics.

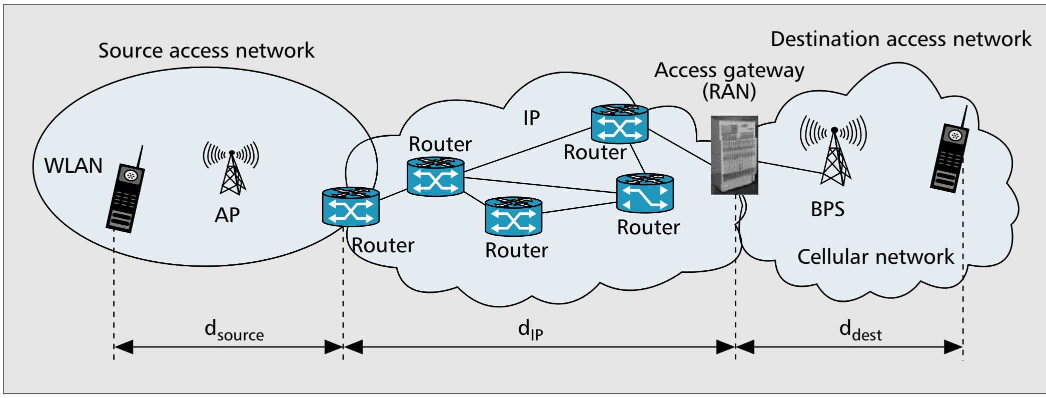

Figure 4. *ETE network delay budget diagram.*

CAC_{L1} — Within every fixed measurement period (e.g., 100 ms):
- The WLAN monitor collects measurements, as mentioned above.
- The admission controller is updated with the most recent measurements from the WLAN monitor.

Whenever a VoIP call initiates from the WLAN:
- The admission controller compares the measured real-time network conditions on MAC process with policies from the admission policy and decides whether to accept or reject the call request based on these policies.
- The admission controller then informs the call agent, the VoIP access gateway, and the router of the level 1 admission controller decisions, accordingly. The call agent will setup the call if the new VoIP call request is accepted in the WLAN.

The most important criterion for the admission controller to admit a new VoIP flow is only when the requested resource is available. An upper bound for bandwidth reservation for VoIP traffic, B_M, needs to be specified. For example, B_M = 60 percent of the bandwidth of a WLAN could be allocated for VoIP calls in a WLAN. The *channel busyness ratio* R_b is defined as the ratio of the time that the channel is determined busy with respect to the total time. Both successful transmissions and collisions contribute to R_b. R_s is denoted as channel utilization, that is, the ratio of successful transmission periods to the total time. R_s counts every period T_{suc} with a successful transmission, and T_{suc} includes the transmission time for RTS, CTS, DATA, and ACK, and all necessary interframe spacings (i.e., SIFS and DIFS). For each VoIP call, it will keep track of the two parameters, *TR* and *len*, in order to characterize the bandwidth requirement of the call, where *TR* is the flow bit rate of the VoIP call (e.g., 10 kb/s), and *len* is the average packet length in bits. The channel utilization *cu* that a flow will occupy is used to describe the bandwidth requirement, $cu = (TR \times T_{suc})/\text{len}$. For the WLAN Voice Manager, the WLAN monitor records the total bandwidth occupied by all admitted VoIP flows in parameter cu_A, which is updated when a real-time flow joins or leaves through the following procedure: When receiving a VoIP connection request from its application layer, a mobile station must send a request with specified cu to the admission controller, noting that it wants to establish a VoIP call. Only after the request is admitted, will the mobile station start to establish the flow with the intended destination, by the function of call agent. Otherwise, the connection request is rejected. Upon receiving a QoS request with parameter *cu*, the admission controller checks whether the remainder of the quota B_M can accommodate the new VoIP call. Specifically, if $cu_A + cu < B_M$, the admission controller accepts the connection request, and updates cu_A accordingly; otherwise, the connection request is rejected. When a VoIP flow ends, the originating mobile station of the flow should send a "*connection terminated*" message to the WLAN monitor, and the latter updates cu_A accordingly. Measurement-based admission control can provide soft delay guarantees for voice traffic [10]. It relies on the measurement period to reflect the dynamics of network status. A shorter measurement period, which means that the measurements are conducted more frequently, can better reflect the networks, but consumes more network resources. A longer measurement period cannot reflect the network dynamics well, but costs less network resources. A good trade-off can usually be determined based on the simulation analysis. An accurate measurement of the network is also very challenging due to the network dynamics.

Once the VoIP calls are accepted, a medium access control (MAC) protocol that is capable of service differentiation and rate control is needed to efficiently coordinate the activities of different types of traffic that coexist in a wireless environment so that the QoS of VoIP calls can be ultimately delivered. The current fundamental access method in the IEEE 802.11 MAC protocol is the distributed coordination function (DCF) [11], which is contention based and is a lack-of-priority mechanism to guarantee the QoS of VoIP packets. In this mechanism, all the stations compete for the channel capacity randomly. The work in [5] proposed a rate control mechanism to dynamically adjust the transmission rate of the coexisting best-effort traffic so that the best-effort traffic would only use the residual bandwidth left by VoIP traffic. The VoIP traffic is given the highest priority in the ongoing queue so that it will have the highest priority to access the channel capacity compared with best-effort traffic. The MAC layer dynami-

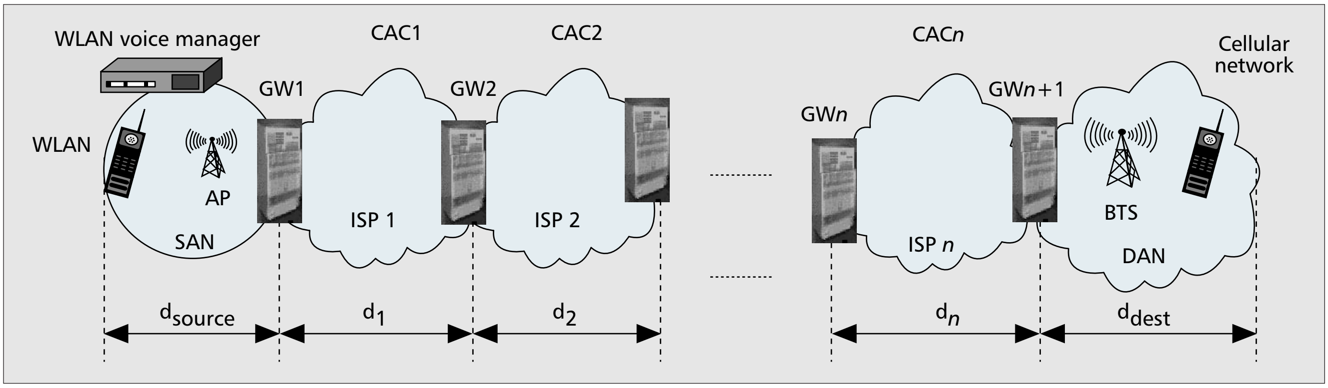

Figure 5. *ETE VoIP over WLAN call admission control.*

cally monitors and updates the free channel capacity, which will be evenly divided among all stations that have best-effort traffic to send. The best-effort traffic will only transmit with the rate given so that the channel capacity will be efficiently utilized and the VoIP QoS will be guaranteed.

Level 2 Solutions — The level 2 WLAN Voice Manager controls the ETE QoS for VoIP calls originated from the WLANs. An ETE CAC algorithm is proposed next. In the ETE CAC algorithm, it is assumed that the standard bandwidth broker and admission controller components are deployed in each of the subnets on the ETE path. The concept of *delay budget* is defined in a subnet as the maximum delay that a packet is allowed within the subnet without loss of quality. For example, if d_{source}, d_{IP}, and d_{dest} are the delay budgets for a Source Access Network (SAN), IP Cloud, and Destination Access Network (DAN), respectively, as shown in Fig. 4, then we obtain the ETE delay budget equal to $d_{source} + d_{IP} + d_{dest}$. Each access gateway needs to convey the delay budget information for its own subnet to the bandwidth broker in the same subnet during network initialization. The delay budget is retrieved by the admission controller in order to perform call admission control. Without loss of generality, assume a VoIP call source is connected to the destination via n ISPs, as shown in Fig. 5. Each ISP, i, has a delay budget, d_i. The SAN and DAN have delay budgets d_{source} and d_{dest}, respectively. The ETE delay budget is $d_{source} + d_1 + d_2 + \ldots + d_n + d_{dest}$. CAC_i stands for the admission controller and bandwidth broker in subnet i; GW_i stands for the gateway between subnet i and subnet $i - 1$. The following ETE CAC algorithm is proposed:

• If d_{source} can be guaranteed in the WLAN SAN through the level 1 WLAN call admission control, a CAC request message is sent from the WLAN QoS Manager through the GW_0 in the SAN to GW_1, which further passes the CAC request to CAC_1. CAC_1 does the following. If d_1 can be guaranteed, CAC_1 sends a local accept decision back to GW_1, which propagates the CAC request to GW_2. If d_1 cannot be guaranteed, CAC_1 sends a reject message back to GW_1, which further informs the Source Access Network to reject the setup request.

• In general, the CAC in subnet i is performed as follows. If d_i can be guaranteed, CAC_i sends a local accept message to QoS GW_i, which further relays the CAC request to GW_{i+1}. GW_{i+1} then passes the CAC request to CAC_{i+1} to do subnet $i+1$ admission control. If d_i can not be guaranteed, CAC_i sends a reject message to GW_i, which in turn informs GW_{i-1} to reject the call setup request. This reject message is then propagated all the way back to the WLAN Source Access Network;

• If d_{dest} can be guaranteed in the DAN, GW in the DAN (GW_{n+1}) passes an accept message back to the source through GW_n, …, GW_1, and the WLAN Source Access Network. Each local call agent starts to do the call setup for its ISP domain, so that an ETE VoIP call can be established. If d_{dest} cannot be guaranteed in the DAN, then the GW_{n+1} passes the reject message all the way back to the WLAN Source Access Network.

In this algorithm, how to determine the delay budget in each subnet is very important. Further research is needed to quantify the delay budget parameter distributions. By proactively using the admission controls for all the subnetworks from source to destination and dynamically provisioning the network, real-time VoIP QoS can be achieved.

For the level 2 Voice Manager, the most important issue is the signaling procedure of the bandwidth broker for real-time ETE QoS control. RSVP [8] was suggested as the major signaling protocol for the bandwidth broker. The bandwidth broker in the SAN domain uses RSVP to request resources from its WLAN Voice Manager. At the WLAN Voice Manager side, the admission controller can make decisions in a distributed manner by the GWs, or centrally by a bandwidth broker. If the GWs are directly involved in the signaling process, they are configured with the corresponding classification, policing, and shaping rules when they grant a request. If a bandwidth broker, rather than the GWs, is involved, the bandwidth broker must configure the GWs when it grants a request.

Conclusions

In this article, a VoIP over WLAN QoS framework, namely, the WLAN Voice Manager, has been proposed. The WLAN Voice Manager can interact with WLAN monitor, call agents, The WLAN Voice Manager can interact with WLAN monitor, call agents, routers, and other VoIP network devices to provision and control VoIP ETE QoS in WLANs. The proposed WLAN Voice Manager consists of five components, which is a logical extension of the standard VoIP call agent.

routers, and other VoIP network devices to provision and control VoIP ETE QoS in WLANs. The proposed WLAN Voice Manager consists of five components, which is a logical extension of the standard VoIP call agent. With different levels of interactions among the five components, VoIP over WLAN QoS can be achieved hierarchically in two levels: level 1 provisioning QoS at the local level through real-time monitoring and measurement-based call admission control and medium access contol, and level 2 ETE QoS guarantee through dynamic resource allocation and parameter-based admission control. The implementation challenges of this framework were discussed, possible solutions were proposed, and some remaining open issues were also addressed.

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BIOGRAPHIES

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