

QoE-based Adaptive mVoIP Service Architecture in SDN Networks

Dongwoo Kwon, Rottanakvong Thay, Hyeonwoo Kim, and Hongtaek Ju

Department of Computer Engineering

Keimyung University

Daegu, Republic of Korea

e-mail: {dwwkwon, vongjacky91, hwkim84, juht}@kmu.ac.kr

Abstract—In this paper, we propose the adaptive Mobile Voice over Internet Protocol (mVoIP) service architecture in Software Defined Networking (SDN) networks to provide the best quality of mVoIP service to end-users. The key challenges in improving the mVoIP Quality of Experience (QoE) are forwarding data path optimization for VoIP flows and optimized codec selection with consideration of network congestion. Based on network Quality of Service (QoS) data and predicted mVoIP QoE data to be collected in SDN networks through mVoIP agents, the proposed service architecture improves mVoIP QoS. In particular, this paper focuses on the improvement of mVoIP QoS by adaptive codec selection optimization and proposes an algorithm for adaptive codec selection in SDN networks.

Keywords—VoIP; mVoIP QoE; Adaptive mVoIP service; Codec selection optimization; Network QoS; SDN

I. INTRODUCTION

Voice over Internet Protocol (VoIP) [1] has become one of the most widely used protocols for voice service delivery. It has been used not only for public services, such as Skype and Viber, but also for private services for companies and organizations. The quality of a VoIP service mostly depends on the voice codec, which is determined by network capability and the number of users to be accommodated. Thus, to improve the quality of a VoIP service, two factors should be considered: VoIP flow control and optimized codec selection.

In Software Defined Networking (SDN) [2] networks, which have provided advances in network management, VoIP flows can be controlled by an OpenFlow [2] controller, such as NOX/POX, Floodlight, Ryu, or OpenDaylight, to guarantee Quality of Service (QoS). To adjust for network congestion, flow tables of OpenFlow switches need to be managed by optimized forwarding path decisions.

Codec selection is another important factor. Due to variation in the number of active users, a fixed voice codec negotiated between a VoIP server and VoIP clients without consideration of the network situation will cause inefficiency and unavailability. In particular, if using a low-quality codec under conditions of spare bandwidth, the utilization of network bandwidth is inefficient; if using a high-quality codec under conditions of insufficient bandwidth, the network bandwidth becomes unavailable. To overcome these limitations, a method of network situation-aware codec selection [3]-[7] is required.

In this paper, we propose the adaptive Mobile VoIP (mVoIP) service architecture based on network QoS data and predicted mVoIP Quality of Experience (QoE) data in SDN networks. The design of our service architecture was guided by two principles: (1) Keep it simple, able to be applied without any modification of VoIP client applications, with an IP Private Branch Exchange (PBX) system, as a practice for VoIP services, (2) Satisfy the Service Level Agreement (SLA) for the VoIP service.

In the proposed architecture, mVoIP QoS is guaranteed by forwarding data path control for VoIP flows and optimized codec selection with consideration of network congestion. To gather QoS data in a wireless network and predict the mVoIP QoE, mVoIP QoE measurement agents are deployed in the wireless network. An mVoIP QoS manager, as a SDN application, collects QoS data on the SDN network from OpenFlow switches and mVoIP QoE data from mVoIP QoE measurement agents. Then, it decides whether the forwarding path for mVoIP flows and the voice codec are optimal under specific network conditions. A selected voice codec is applied by an mVoIP QoS adapting agent within an IP PBX system in the SDN network.

This paper focuses on adaptive codec selection followed by the network QoS and mVoIP QoE data, while the forwarding path decision in SDN networks depends on an OpenFlow controller and a network slicer such as FlowVisor [8]. The rest of this paper is organized as follows. Related studies are discussed in Section II. In Section III, the adaptive mVoIP service architecture in SDN networks is described in detail, including the mVoIP QoS manager, the mVoIP QoS adapting agent, and the mVoIP QoE measurement agent. The algorithm for adaptive codec selection in SDN networks is also proposed. In Section IV, the conclusions are presented and future works are discussed.

II. RELATED WORK

Takahashi et al. [9] presented the factors that determine QoS for a VoIP service and a subjective/objective quality assessment. For subjective quality assessment, opinion rating and the opinion equivalent-Q method are introduced. Opinion models such as E-Model and speech-layer objective models such as Perceptual Evaluation of Speech Quality (PESQ), P.563, and P.AAM are introduced for objective quality assessment. Packet-layer objective models such as P.VTQ are also introduced for objective quality assessment, which is based merely on IP packet information except for

speech data in the payload. To investigate the relationship between subjective and estimated quality, the experiment using the modified E-Model was conducted, and the results were presented. Finally, a framework for the development of quality assessment research was proposed.

Kim et al. [10] measured PESQ scores to compare the performance of various codecs, such as G.711, GSM, iLBC, Speex, and Skype's codec, in wireless networks. The experiments were performed by transmitting audio signals from one VoIP client to another using the Jack audio router through the NIST Net emulator to measure the performance of the VoIP codecs. As a result of these experiments, these researchers found that Skype was the most robust against packet delay and packet loss. Among codecs other than commercial services, Speex was robust when packet loss was less than 10% and iLBC was very robust under network congestion. These results can be utilized to initialize default codec sets for the adaptive codec selection method that we propose in this paper.

Sfairopoulou et al. [3] evaluated the performance of a combined Call Admission Control with Codec Selection (CAC/CS) algorithm proposed in [4], based on three policies: non-adaptive policies that start with the lowest and highest bandwidth codecs, and an adaptive policy that changes codec in randomly active calls. In particular, experiments were conducted applying the adaptive policy to three cases: applying codec selection (1) only to new call requests (*on new*), (2) only in the presence of rate changes (*on rate*), or (3) both for new calls and at any rate change (*on both*). The *on new* case presented a low blocking probability but high dropping probability; the *on rate* case maintained high estimated Mean Opinion Score (MOS) values under high traffic load; and the *on both* case presented a trade-off between blocking and dropping probabilities but the worst estimated MOS values. In our study, the proposed architecture involves controlling VoIP flow paths according to network congestion to guarantee a predefined Call Setup Success Rate (CSSR) and Call Drop Rate (CDR), comparing them with the CSSR and CDR obtained from an IP PBX system. Also, the codec selection method considers the number of active calls and the movement of each VoIP client.

Ng et al. [5] proposed and evaluated an adaptive codec switching algorithm for VoIP applications in wired and wireless networks. Codec switching during active calling is achieved by renegotiating the audio session using a RE-INVITE message in Session Description Protocol (SDP). To evaluate the proposed algorithm, a Session Initiation Protocol (SIP) proxy server was deployed on the real network and SIP clients were connected to the wireless Access Point (AP). To simulate network congestion, the network traffic emulator was also deployed on the wireless network. The result was effective in increasing the average MOS values when switching between PCMU and GSM, while the packet loss rate was more than 16%. However, only the packet loss rate was considered to adaptively determine the optimized codec in this study. To apply the proposed algorithm, SIP client applications should be modified based on the calculated packet loss rate. In our

proposed architecture, both a client application and an IP PBX system require no modification to be practical.

Roychoudhuri et al. [6] proposed adaptive rate control for audio packets, which is based on packet loss prediction and on-line audio quality assessment. Audio Genome was used to store the codec type, loss distribution, and delay to derive the audio quality of the ongoing transmission. The adaptive rate control framework was proposed to combine all audio codecs with Audio Genome to maintain optimized audio quality. In the proposed framework, the Rate-Quality Optimization problem is to maximize the audio quality under the constraints of available bandwidth and delay. The experiment for Rate-Quality Optimization was conducted with the codecs PCMU, G.721, GSM FR, G.728, and G.723.1 in six scenarios. In the scenario of low available bandwidth and low delay among them, the feasible solution was G.728 (61%) and PCMU (39%). In the worst-case scenario of low available bandwidth and high delay, the feasible solution was G.721 (72%), G.728 (19%), and PCMU (8%). One of our approaches is to determine the priority set of optimized codecs and quality options, when VoIP client applications and an IP PBX system are used without any modification. Thus, these results can be used to create default codec sets in our adaptive codec selection method with [10].

Qiao et al. [7] proposed a QoS control scheme that combines rate-adaptive and priority marking QoS control to improve speech quality. A VoIP simulation system with a NS-2 network simulator was used to simulate VoIP flow, which includes an encoder/decoder/marker for the Adaptive Multi-Rate (AMR) codec, a bitrate controller, and a loss simulator. The MOS scores were predicted using PESQ based on given AMR rates and packet loss. Because each VoIP client application supports different codecs, a set of diverse codecs including non-multi-rate codecs is applied in our adaptive codec selection method.

III. ADAPTIVE MVOIP SERVICE ARCHITECTURE

The quality of VoIP service depends on the codec used and the network condition. If high-quality codecs are used for the VoIP service, the voice quality is very good, but the number of VoIP channels is less than in the case of using low-quality codecs due to VoIP bandwidth consumption.

The bandwidth consumption, which is represented by B , can be calculated by (1). The codec bit rate and packets per second (PPS) are calculated by dividing the codec sample size (SS) by the codec sample interval (SI) and dividing the codec bit rate by the voice payload size (PS), respectively. The total bandwidth consumption for the VoIP service can be calculated by multiplying the VoIP packet size (P) by PPS and the number of channels (C), while P is represented by the sum of data link layer header, the IP/UDP/RTP header, and the voice payload size.

$$B = (SS \times P \times C) / (SI \times PS) \quad (1)$$

In SDN networks, OpenFlow switches maintain their flow tables based on the FlowModify messages which are sent from the OpenFlow controller to determine the shortest

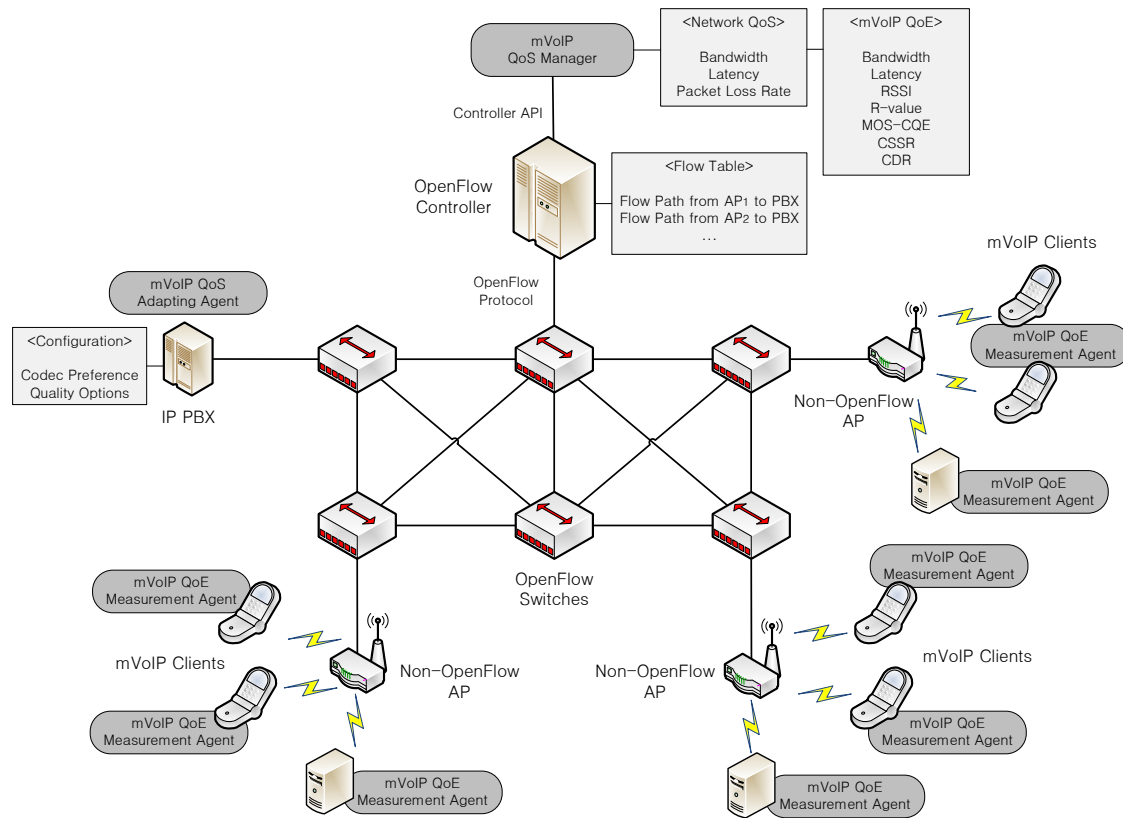


Figure 1. Adaptive mVoIP Service Architecture in SDN Networks

packet paths. To improve and guarantee VoIP QoS, the flow paths between the IP PBX system and the VoIP clients should be determined by consideration of the network conditions, such as bandwidth and latency. After that, the optimized codec can be selected based on the bandwidth usage rate and the channel requirement. It allows the priority of the codec to be rearranged based on the network condition and the number of currently active users of the VoIP service.

This paper proposes an adaptive mVoIP service architecture in SDN networks, which consists of the mVoIP QoS manager, mVoIP QoE measurement agents, and the mVoIP QoS adapting agent as shown in Fig. 1. The SDN network connects with an OpenFlow controller, OpenFlow switches, Non-OpenFlow wireless APs for connecting to mVoIP clients, and an IP PBX system.

A. mVoIP QoE Measurement Agent

In the proposed architecture, mVoIP QoE measurement agents on the dedicated systems conduct bandwidth and latency measurements from themselves to the IP PBX system. Although the controller has all the information on the network topology and can measure bandwidth and latency by itself, the agents need to measure them because non-OpenFlow APs are regarded as edge nodes in SDN networks, and the links between an AP and its connected devices may be critical bottlenecks. These measurements are not conducted by the agents on mobile devices due to excessive battery consumption.

The agent on a mobile device merely collects the Received Signal Strength Indication (RSSI) value from broadcasting information, sent by an AP, to judge the mobility of the VoIP clients. The movement of the VoIP client can be judged by GPS, gyroscope, and RSSI data. Because the GPS or the gyroscope sensor of a mobile device deals with movement in very wide or narrow areas, respectively, the RSSI value is used to judge the mobility of the device. It also directly reflects the wireless network condition. Thus, frequent variation of the RSSI value shows that a VoIP client is moving and implies that a codec with low bandwidth consumption is determined in spite of sufficient available bandwidth.

The VoIP quality metrics are categorized as call setup quality and call quality. Call setup quality is estimated by CSSR and CDR, which can be gathered from the mVoIP QoS adapting agent that communicates with the IP PBX system. Low CSSR and CDR show that the flow path between the AP and the IP PBX system should be modified to avoid network congestion.

Call quality can be measured by a subjective MOS test. Due to time-consuming efforts and expensive cost, the E-Model is used as an alternative to human-based MOS estimation to predict call quality. The gross score, which is represented by the *R*-value, of the E-Model is computed by:

$$R = R_o - I_s - I_d - I_e + A \quad (2)$$

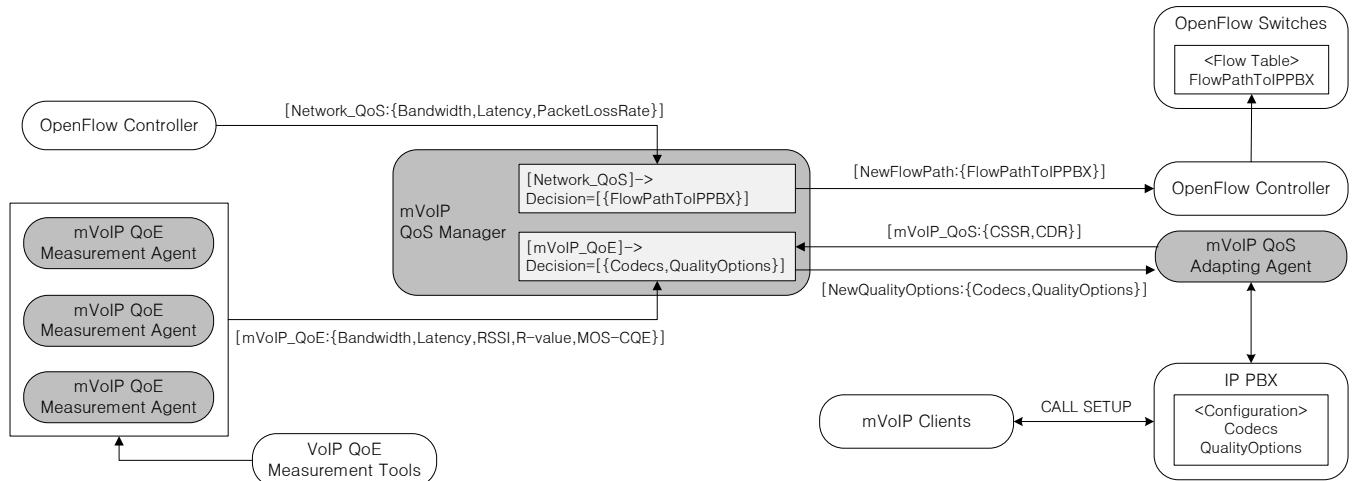


Figure 2. mVoIP QoS Manager

where R_0 represents the basic signal-to-noise ratio; I_s represents all impairments that occur simultaneously with the voice signal; I_d represents all impairments caused by delay and echo effects; I_e represents impairments caused by low bit-rate codecs; and A is an advantage factor that allows for an advantage of access. R -value can be translated to MOS-Conversational Quality Estimated (MOS-CQE) by (3) [11],[12].

$$\begin{aligned} \text{MOS-CQE} = & \quad (3) \\ & 1 \quad (R < 0) \\ & 1 + 0.035R + R(R-60)(100-R) \cdot 7 \cdot 10^{-6} \quad (0 < R < 100) \\ & 4.5 \quad (R > 100) \end{aligned}$$

R -value is calculated by using VoIP QoS measurement tools, such as VoIPmonitor [13] and pjsip-perf [14], and is transformed into a MOS scale by the agents. Then, the agents send all the gathered data to the mVoIP QoS manager, including the bandwidth, the latency, and the RSSI data, to adaptively select the optimized codec.

B. mVoIP QoS Manager

Based on the network QoS and estimated mVoIP QoS information, such as MOS-CQE, the mVoIP QoS manager determines the flow path between an AP and the IP PBX system and chooses the codec preference. The new flow path is updated by the OpenFlow controller, and the new codec preference is set by the mVoIP QoS adapting agent into the IP PBX system. Fig. 2 depicts data flow among the manager and agents on the proposed architecture.

The algorithm for adaptive codec selection according to network conditions and VoIP service policies is described in Table I. First of all, before determining an optimized codec preference set, the flow paths among the APs and the IP PBX system are recalculated, particularly when CSSR or CDR does not satisfy the predefined threshold. For example, in Korea, the CSSR and CDR thresholds can be defined as 95% and 5%, respectively, according to the standard proposed by Telecommunication Technology Association (TTA) [15].

In the proposed algorithm, the codec set consuming minimal bandwidth is selected when the number of current active users is greater than the designed channel requirement, the network latency is lower than the predefined latency threshold, or the mVoIP client is moving. A latency threshold can be defined at 150ms, as proposed by TTA [15].

For premium users, the codecs providing the best voice quality, such as PCM, under the available bandwidth are selected when not group-calling. For the other users, the codec with the highest MOS-CQE in the minimal bandwidth set among the default sets of the APs is set if the consumed bandwidth of the selected codec is under the available bandwidth and the average bandwidth-consuming current available channel with extra one channel is also under the available bandwidth.

TABLE I. PROPOSED ADAPTIVE CODEC SELECTION METHOD

Algorithm: Adaptive Codec Selection for mVoIP in SDN Networks	
input	Caller: C_r Set of Callees: $CE = \{ CE_1, CE_2, \dots, CE_n \}$ AP set: $AP = \{ AP_1, AP_2, \dots, AP_n \}$ Available bandwidth set of each AP: $B = \{ B_1, B_2, \dots, B_n \}$ Latency set of each AP: $L = \{ L_1, L_2, \dots, L_n \}$ Set of default codec set of each AP according to channel requirements: $DEFset = \{ DEFset_1, DEFset_2, \dots, DEFset_n \}$ Set of average variation per unit time of RSSI of each client connected to the specific AP: $R = \{ R_1, R_2, \dots, R_n \}$ MOS-CQE set of each AP: $MOS = \{ MOS_1, MOS_2, \dots, MOS_n \}$ CSSR set of each AP: $CSSR = \{ CSSR_1, CSSR_2, \dots, CSSR_n \}$ CDR set of each AP: $CDR = \{ CDR_1, CDR_2, \dots, CDR_n \}$ Supported codec set (sorted by low bandwidth and high packet interval): $C = \{ C_1, C_2, \dots, C_n \}$ Required channel number set of each AP: $REQ = \{ REQ_1, REQ_2, \dots, REQ_n \}$ Active user set in IP PBX: $USER = \{ USER_1, \dots, USER_n \}$ Premium user set: $PUSER = \{ PUSER_1, PUSER_2, \dots, PUSER_n \}$ Predefined CSSR threshold set of each AP: $CSSRTHR = \{ CSSRTHR_1, CSSRTHR_2, \dots, CSSRTHR_n \}$ Predefined CDR threshold set of each AP: $CDRTHR = \{ CDRTHR_1, CDRTHR_2, \dots, CDRTHR_n \}$ Predefined latency threshold set of each AP: $LTTHR = \{ LTTHR_1, LTTHR_2, \dots, LTTHR_n \}$ Predefined RSSI variation value to infer whether a client is moving: $MRssi$

output	Priority set of selected codecs: $S = \{ S_1, S_2, \dots, S_n \}$ Quality option set of selected codecs: $Q = \{ Q_1, Q_2, \dots, Q_n \}$
1	$S \leftarrow \{ \}$
2	$Q \leftarrow \{ \}$
3	$apcr \leftarrow AP_{i \in \{Cr\}}$
4	$apce \leftarrow \{ AP_{i \in \{CE1\}}, AP_{j \in \{CE2\}}, \dots, AP_{k \in \{CEn\}} \}$
5	$ap \leftarrow apcr \cup apce$
6	if $CSSR_{ap} < CSSR_{THR_{ap}}$ or $CDR_{ap} > CDR_{THR_{ap}}$:
7	Request to the controller to assign the optimized flow path between the APs, which have low CSSR or high CDR, and the IP PBX system
8	if $NewFlowPath_{(NODE_{ap} \rightarrow NODE_{pbx})} = PreviousFlowPath_{(NODE_{ap} \rightarrow NODE_{pbx})}$:
9	$DEF_{min} \leftarrow \min_{bandwidth}(DEF_{set_{ap}})$
10	$S \leftarrow DEF_{min}$
11	$Q \leftarrow DEF_{min}.option$
12	return
13	end if
14	end if
15	if $NUM_{USERi \in ap} > REQ_{ap}$ or $L_{ap} > LTTHR_{ap}$ or $(apcr_{RCr} \geq MR_{ssi}$ or $apce_{RCr} \geq MR_{ssi})$:
16	$C_i \leftarrow \max_{MOS-CQE}(C_1, \max_{MOS-CQE}(\{ C_i \text{ set} \mid 2.6 \leq C_i.mos-cqe \leq \min(MOS_{ap}) \}))$
17	$S \leftarrow \{ C_i, C_{i-1}, \dots, C_1 \}$
18	$Q \leftarrow \{ C_i.option, C_{i-1}.option, \dots, C_1.option \}$
19	else :
20	if $NUM_{CE} = 1$ and $(Cr \in PUSER \text{ or } CE_1 \in PUSER)$:
21	$C_i \leftarrow \max_{bandwidth}(C_1, \max_{MOS-CQE}(\{ C_i \text{ set} \mid C_i.bandwidth \leq \min(B_{apcr}, B_{apce1}) \}))$
22	$S \leftarrow \{ C_i, C_{i-1}, \dots, C_1 \}$
23	$Q \leftarrow \{ C_i.option, C_{i-1}.option, \dots, C_1.option \}$
24	else :
25	$DEF_{min} \leftarrow \min_{bandwidth}(DEF_{set_{ap}})$
26	$bw_{mos} \leftarrow \max_{MOS-CQE}(DEF_{min}).bandwidth$
27	$bw_{avg} \leftarrow \text{avg}_{bandwidth}(DEF_{min})$
28	if $bw_{mos} < \min(B_{ap})$ and $(\min(REQ_{ap} - NUM_{USERi \in ap}) + 1) \times bw_{avg} \leq \min(B_{ap})$:
29	$DEF_{mos} \leftarrow \text{SORT}_{MOS-CQE, Desc}(DEF_{min})$
30	$S \leftarrow DEF_{mos}$
31	$Q \leftarrow DEF_{mos}.option$
32	else :
33	$S \leftarrow DEF_{min}$
34	$Q \leftarrow DEF_{min}.option$
35	end if
36	end if
37	end if

C. mVoIP QoS Adapting Agent

The mVoIP QoS adapting agent applies the new codec preference and quality options to the IP PBX system. Fig. 3 shows the interaction between the agent and the IP PBX system in detail. In Fig. 3, Asterisk [16] is used as an IP PBX server, as it is the most popular IP PBX. The agent communicates with Asterisk Manager Interface (AMI) to maintain the action table.

The mVoIP QoS manager gathers information, such as the supported codec list, CSSR, and CDR, from the IP PBX system and updates the codec priority of each VoIP user and codec quality options to the IP PBX system through the mVoIP QoS adapting agent.

IV. CONCLUDING REMARKS AND FUTURE WORK

This paper discusses previous studies of VoIP service and its performance in detail and proposes the adaptive mVoIP service architecture based on network QoS and mVoIP QoE data in SDN networks to improve mVoIP QoS. In this architecture, two key approaches are proposed: flow path optimization for mVoIP traffic using the SDN controller and the adaptive codec selection method. In particular, this paper focuses on a network condition-aware codec selection method and proposes an algorithm for adaptive codec selection for mVoIP in SDN networks.

In the future, the performance of the proposed algorithm will be evaluated in real networks and flow path optimization according to the network condition, which is the other approach, will be studied between APs and an IP PBX system in SDN networks.

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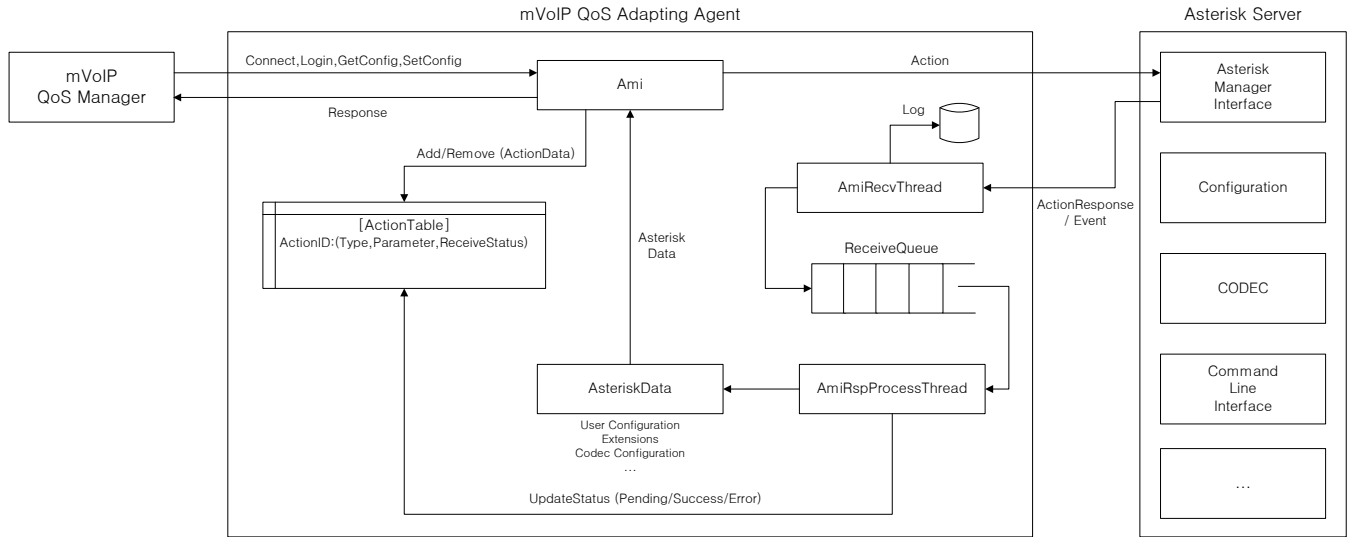


Figure 3. mVoIP QoS Adapting Agent

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