

## SEAMLESS VOIP SERVICE USING CENTRALIZED CODEC ADJUSTMENT: A FRAMEWORK IN WIRELESS NETWORK

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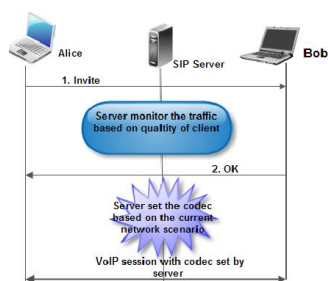
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### Graphical abstract



### Abstract

The evolution of wireless technology leads VoIP application to be a good preferred choice among mobile users. Customers can having a VoIP service while being mobile regardless of location, terminal, and it is independent of network access. It is believed to be an alternative to the PSTN and its service is expected to be as good as PSTN. In essence, QoS plays a crucial part in order to maintain the seamless service during the VoIP session. As wireless networks are prone to channel error and VoIP application is known to be a delay-sensitive application, network traffic congestion could easily affect the voice quality of VoIP. Therefore, the codec selection within the VoIP application must be improved so that the codec can operate at various bit rates in order to cater for different traffic conditions. In this paper, we explain how centralize codec switching can help user to communicate seamlessly. We use WiFi network traffic as an example scenario.

**Keywords:** Codec adjustment, centralized codec adjustment

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## 1.0 INTRODUCTION

VoIP continues to be the popular application for fixed-line and mobile users. With various wireless access technologies such as GPRS, WiFi, WiMAX and 3G, VoIP can be experienced while being mobile regardless of location, terminal, and it is independent of network access. It is a possible alternative to the Public Switch Telephone Network (PSTN) and its service is expected to be as good as PSTN. However, VoIP is delay-sensitive application and traffic congestion could easily affect the performance of VoIP sessions. Various factors such as capacity, bandwidth and network topology acts as an important role in contributing to traffic congestion.

In wireless communication, elements such as volume of calls, type of services and network architecture affect the traffic congestion [1]. Thus, VoIP traffic requires minimum Quality of Services (QoS) guarantee from the network to meet its stringent bandwidth, delay and jitter requirement [2]. QoS must be maintained across wireless networks to ensure that delay-sensitive VoIP receives consistent and high

priority end-to-end communication to provide optimal call quality. For VoIP users, QoS is all about guaranteeing that voice traffic will not be delayed or dropped due to network congestion. VoIP transmits voice data packets in compressed form done by Codec (COder-DECoder, also known as encoder/decoder or Compresion-decompression) [3] so that the transmission is lighter, thus improving the service performance.

One of the issues that need to be emphasized in the deployment of VoIP is the availability of bandwidth in which it affects the voice codec and when/where to use compression in order to meet a service goals [4]. Therefore, it is important to provide a good codec adjustment structure, and this can be attained by having the ability to change the codec at multiple rates on different traffic conditions. Codec adjustment can be adapted to retain voice quality to users [5]. Through codec adjustment, users can still continue an audio communication without terminating a session. In fact, even the users will not be aware of the changes of the codec. The selection of suitable codec is useful

for efficient bandwidth utilization. However, adjustment codecs available in the literature only perform codec switching on terminal clients. This method could result in the unfair selection of codecs among users of high network traffic, whereby during high traffic, network bandwidth could not support many clients. This caused some users to face with severe audio quality as higher codec is being conquered by a single user. Hence, we propose a method in which the codec adjustment is done centrally, subject to the availability of bandwidth. Through this method, the server will check the network bandwidth availability and send the notification to the clients the suitable codec and preserve the connection for the current traffic network. This will provide an optimum video quality to all clients in the same network and codec selection is evenly distributed among clients. In this paper, there are two challenges that are addressed in order to support efficient and effective adjustment:

- The adjustment must be intelligently triggered to ensure that an application-specific QoS is not significantly affected.
- Determine which type of adjustment to take place at different conditions, i.e. considering the network traffic status as well as the user/application preferences.

In this paper, we describe a centralized codec adjustment procedure for VoIP under traffic congestion to maintain good VoIP session calls. This is a preliminary study for further research on centralized codec adjustment in VoIP application. The remainder of this paper is organized as follows. Sections two discusses the framework overview, section three explains the framework implementation and finally a conclusion in section four.

## 2.0 FRAMEWORK OVERVIEW

### 2.1 Codec Adjustment Scenario

Figure 1 is the illustration on how a centralized codec adjustment can be adapted. Alice is having a conversation with Bob using a laptop which is equipped with WiFi connection. At this time, consider high bit rate codec is used. As there are a number of people accessing the WiFi connection at the same time, so the network bandwidth consumption increases. This conditions results to a degrading voice quality, hence the voice conversation between Alice

and Bob will be affected. However, this condition can be avoided by having a centralized codec switching mechanism in the server that detects the condition of high traffic, thus the server will choose the most appropriate codecs to be used at the time. Unnoticed by Alice, her laptop improves the conversation by accepting the changing of the codec to a low bit rate codec that consumes less bandwidth yet provide an acceptable voice quality. As a result, the conversation is maintained without any degradation and disruption. Finally, when the load on the access point is reduced, the demand for bandwidth will reduce accordingly.

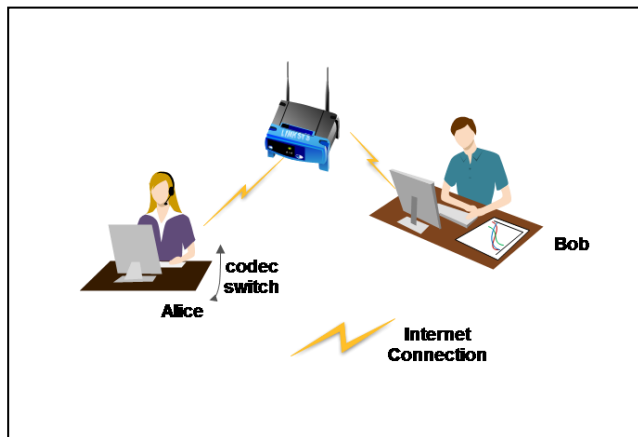


Figure 1 Block diagram of the processes of the system

There are a number of codec used in voice application as listed by Table 1 that shows the most important encoding standards covered by the International Telecommunications Union (ITU)[6]. Different coding techniques for telephony and voice packets are standardized by the ITU-T such as G.723.1, G.729, G.729A and etc [7].

### 2.1 SIP Flow Process

In our framework, we use Session Initiation Protocol (SIP) [8, 9] for signaling protocol. Figure 2, depicts the SIP message flow when the session establishes and the changing of codec during the realtime communication. The online session between Alice and Bob can be explained as follows:

- \* Alice invite Bob to have a VoIP conversation.
- \* Server will monitor the current traffic.
- \* Server sends notification to client (Alice) on the best codec to be used at that time.

Table 1 ITU Encoding Standards

ITU Standard	Description	Bandwidth (Kbps)	Conversion delay (ms)
G.711	PCM	64	<1.00
G.721	ADPCM	32, 16, 24, 40	<1.00
G.728	LD-CELP	16	~2.50
G.729	CS-ACELP	8	~15.00

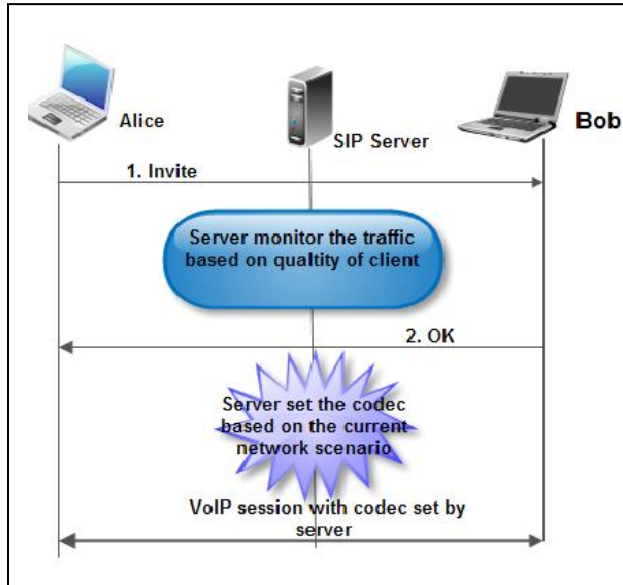


Figure 2 SIP flow of centralized codec adjustment

### 2.2 QoS Parameters

There are a several of QoS parameters can be identified as listed in Table 2 [10]. However, in our preliminary experiment, packet loss is selected as a parameter for QoS for the codec. Packet loss is defined as a failure of packet to arrive at the destination [11]. Other factors that contribute to packet loss: inadequate signal strength at the destination, natural or human-made interference, excessive system noise, hardware failure, software corruption or overburdened network nodes [12]. Percentage of packet loss rate,  $y$ , can be calculated by the equation as below:

$$y = \frac{a}{b} \times 100 \tag{1}$$

where:

$a$  = number of packet loss

$b$  = the number of RTP packets being transmitted.

Table 2 Network QoS Parameter

Category	Parameters
Timeliness	Delay, Response Time, Jitter
Bandwidth	Systems-level data rate, Application-level data rate, Transaction time
Reliability	Mean time to failure (MTTF), Mean time to repair (MTTR), Mean time between failures (MTBF), Percentage of time available, Packet loss rate, Bit error

## 3.0 FRAMEWORK IMPLEMENTATION

### 3.1 Experiment Setup

The setup of the propose architecture as depicted in Figure 3 consists of:

- WiFi access point (AP) mobile access network.
- 3 laptops: 2 SIP clients as Alice and Bob, and another laptop was configured as a traffic client.
- 2 machine servers for SIP server and Traffic Generator server. The SIP server also was equipped with a codec adjustment mechanism to monitor the network traffic.
- Run SIP clients, Alice and Bob by sending and receiving SIP messages to each other.
- After some point, run traffic generator to increase traffic, in order to experience the voice quality before and after the high traffic.

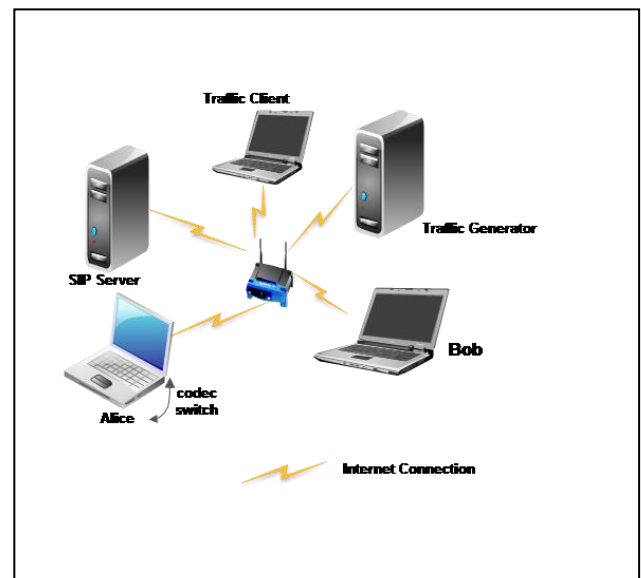


Figure 3 SIP flow of centralized codec adjustment

In Figure 3, clients (Alice and Bob) on laptop are installed with SIP-based VoIP application that enables them with VoIP communication as well as codec adjustment. All of them are connected to the WiFi. All clients will have to register themselves with the SIP server in order to communicate to each other. Alice will be communicating with Bob by sending and receiving SIP signaling messages between SIP server and clients. Two audio codec with different bit rate are used in the experiment-PCMU-128kbps and Speex-44.2kbps. The traffic generator is a traffic simulator meant to send packet to the clients in order to increase traffic.

The sending and receiving packets is gathered by RTP RR [13] which is a kind of RTCP message information collection at a certain time interval for QoS such as delay, jitter, packet. RTP RR data later

will be analyzed to calculate percentage of packet loss at that time. If the packet loss percentage is below or equal to the threshold value, the codec system will triggered to the high bit rate codec. During the heavy traffic, ongoing VoIP session will be experience bad QoS and percentage of packet loss exceeds the threshold. Hence, Alice will switch a codec from high bit rate to the low bit rate, and later on will trigger to the high bit rate codec if the network traffic is less congested.

### 3.2 Testing Scenario and Result

Test plan scenario will take place as follows:

- Run SIP server as well as codec adjustment mechanism
- Run SIP clients, Alice and Bob by sending and receiving SIP messages to each other.
- After some point, run the Traffic Generator to increase traffic, in order to experience the voice quality before and after the high traffic.

The system was run and executed for more than 20 times to get the consistent result. At the point where the packet rate from the traffic generator is more than the threshold, the codec is switched to the low bit rate, GSM and otherwise it will be switched to PCMU. During the experiment, users are asked to listen to the quality of the audio before and after the traffic congestion. Mean Opinion Score (MOS) was used to measure the quality of audio in our experiment. Table 3 shows the satisfaction level of the end users [14].

**Table 3** MOS: User Satisfaction Level

MOS	Traffic Size (Number of users)
5	Perfect (Like face-to-face conversation or radio reception)
4	Fair. Imperfections can be perceived, but sound still clear. This is (supposedly) the range for cell phones.
3	Annoying.
2	Very annoying. Nearly impossible to communicate.
1	Impossible to communicate

**Table 4** User MOS Rating without adjustment

Packet Loss (%)	MOS	Traffic Size (Number of users)		
		10	20	30
1	Good	3	2	2
3	Fair	3	1	1
6	Poor	0	1	3
7	Bad	0	1	1

**Table 5** User MOS Rating with code adjustment

Packet loss (%)	MOS	Traffic Size (Number of users)		
		10	20	30
1	1 Good	0	0	0
3	3 Fair	3	3	3
6	6 Poor	0	0	0
7	7 Bad	0	0	0

MOS test was performed with and without codec adjustment. Table 4 and 5 shows the MOS rating for different traffic condition. Table 4 shows most users rate the "fair" value during high traffic, showing the codec was adjusted at the same rate to all users during high traffic, as opposed to Table 3 which explains the unfair codec used during high traffic [15].

## 4.0 CONCLUSION

As the availability of bandwidth is crucial in VoIP services and the bandwidth consumption typically relies on the codec used, there is a need to adjust the codec to match the surging network traffic at that time. In this paper we present a switching scheme that can help VoIP user to stay connected on their call while in traffic congestion. In this case we use traffic congestion in WiFi network as a scenario. The work is an initial stage of studying QoS for VoIP in WiFi network environment. Future work will include performance testing and analysis using simulation in order to evaluate the performance of our implementation. We will also increase the parameter of QoS in our future research such as jitter and delay.

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