Wireless based Solution for VoIP over the campus of Moncton University

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Abstract

It is no doubt that the Internet has deeply changed our life. We depend on it more and more no matter for our work and our home. Following this trend, nearly every application associated to the Internet has gained great popularity as soon as their appearance, include VoIP. As a tool of communication, it's well known of its convenience and its ability of reducing telephone expense. Nowadays, the combination of the VoIP and wireless network, which permit us to access to the Internet whenever and wherever, has become a strong rival for the traditional telephony. In this paper, we present wireless based software designed to adopt a specific network environment. The developed solution is for the use over the campus of Moncton University without needing or using a distant server.

1 Introduction

From its name, we can understand easily that VoIP is an application based on the IP network. With using encoding and decoding, the signals of the voice could be transmitted on the Internet in the form of the packets. As we use just one network to deal with two kinds of streams instead of using traditional separate data and voice networks, it can save much more cost of the service. The form of VoIP varies, such as PC to phone, PC to PC or phone to phone. [1]

The system of the VoIP is constituted by the access terminals, gateways and gatekeepers of the both callers and receivers [2]. The core content is the gateway, which lies on the application layer and is used to realize the connection between the networks like IP-based network and the PSTN network. It can make sure the IP address of the receiver by the information in the basement of the telephone numbers. And then, it will add the data of the IP address in the package and choose a best path to reduce the delay. Then the package of the voice data can arrive to the gateway of receiver over the Internet. The principle works of the gatekeeper are the authentication of the

users, to parse the address, to manage the secure and so on.

Generally, the VoIP is based on the packet switching network of the Internet, which is different from the traditional telephony network that is based on the circuit-switch. So it can use the UDP protocol to achieve transmissions after a series of disposals like compression and packaging the analog voice data into the form which can be transmitted over the IP based network. The process of the VoIP can be divided into several steps contains the switch from the voice to the data, the packetization of the data, the transmission, unpackitization and return from the digital data to the analog voice. [3]

The system structure of the VoIP is shown in the Figure 1.

Figure 1. Structure of the VoIP system

2 State-of-the-art

The foundation of the technology is of sorts protocols. The three main protocols used in the VoIP are H.323, SIP, MGCP.

2.1 Session Initiation Protocol (SIP)

The SIP is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. It can be used to create two-party, multiparty, or multicast sessions that include Internet telephone calls, multimedia distributions, and multimedia conferences. (RFC 3261). It can also be used to invite a part to participate into an existed session which is initiated in the other ways. The SIP is designed originally for the voice and video calls.[4]

The SIP doesn't define the type or the content of the sessions, but it defines how to control the sessions and is concerned about the arrival to the destination of the conversation. That provides us a much better flexibility which the SIP has a larger scale of application like voice call, video call, alternating game and web conference. It's easier to read and debug because it's based on the text. It's also intuitionistic for the designer.

SIP is able to use any IP based network and any IP equipment. So its ability of combining these resources can bring us more opportunity to develop the technology.

2.2 H.323 protocol

H.323, is a family of the protocols which contains H.323, H.263, H.723.1, H.245 and H.225. H.323 is a set pattern that involves terminal equipments, video transmission, voice transmission, data transmission, communication controlling and network interface. Also it consists with the contents that contribute the multi-point conference like multi-point control unit, multi-point controller, multi-point processor, gateway and gatekeeper. [5] It has been designed for the telecommunication, and it's proper for constructing the larger scale network. With using H.323, it's easy to attain the ability of controlling and co-operability among the equipments of different providers.

Nowadays, the most popular directions of the VoIP's development are the H.323 and SIP. Both of them are based on the package-switch network and have the complete design for the VoIP system. However, according the way of the designing and communication, the ability and some other aspects of

the performance, they all have some inimitable advantages. [6]

H.323 is much older than the SIP, so it's more mature than the SIP. But it's also more complex than the latter. H.323 is designed by the ITU-T, it supposes that the VoIP is just an equivalent of the traditional telephony with some little difference, such as the way of the transmission, which is that the VoIP uses the package-switch while the traditional telephony uses the circuit-switch. And also, the VoIP transmits the digital data and the object of the latter is the simulate data. But for the SIP, it considers the VoIP just as an application over the Internet with the request of the QoS and protocols. Because of the way of designing, the H.323 is easier to connect the telephony network and the Internet which supports VoIP. While the most prominent advantage of SIP is that it has a strong ability of extending. In another word, H.323 is a concentrative system which is under the control of the gatekeeper, so it's a good candidate for a big scale VoIP network. The service providers can easily deal with the account of the charge and the controlling of the bandwidth. Although SIP lacks the experience to undertake the huge job like taking on a large scale network to serve the VoIP, it's more flexible for the application over the Internet and for the service provider to work out more function goes with the VoIP due to its simplemnt. [7]

The NetMeeting of the Microsoft is a typical represent of the usage of the H.323 whereas the MSN messenger is a SIP system.

2.3 Real-time transportation protocol (RTP)

The technology of the real-time transmission depends mainly on the RTP (real-time transport protocol) which provides the transmission pear-to-pear for the real-time data includes voice and video. It is based on the datagram protocols. [8] RTP contains two parts, data and controlling. It has the time label and it controls synchronization of the different data streams to help the receiver reform the data packages which are got from the sender. RTP doesn't have any guarantee of the quality of service or of the transmission on schedule. It relies on the lower layers and RTCP to complete these goals. RTCP (real-time transport control protocol) is a co-agent of RTP when a session is on going. During the conversation, all the participants should send a package of RTCP every certain time which contains the statistic of the amount of the sent packages, lost packages and so on. It is responsible on the dealing with the feedback information, providing RTP a permanent name as a symbol in the transportation layer and also, it will

adjust the sending rate of the RTP packages for every participant of the session according to the number of the users. [8] With the collaboration of RTP and RTCP, the server can predominate dynamically the situation in the network to adjust the data rate. So we can optimize the data rate with the least overhead.

3 Wireless based solution over the campus

Originally, we use more the service based on the wire line network. After the appearance of the mobile technology like cellular network, the portability of the network has been much developed. But, as we know that the cellular network is good at the transmission of the voice, it's difficult for it to deal with a mass of the data work. [9] So appear wireless network that can solve the conflicts between the portability and the data work simultaneously.

According as the coverage, wireless network can be classified into different kinds including WLAN (wireless local area network), WMAN (wireless metropolitan area network) and WWAN (wireless wide area network). Generally, the WLAN is used in the office or the home environment because its coverage scale is about hundred meters. The classified of the wireless network is shown in the figure 3 below. And WMAN can be used in the whole area like a city, while WWAN has a larger coverage. In our daily life, we will use more often WLAN as its coverage is enough for our home or work. The most popular standard that used for the WLAN is the IEEE802.11 family which is a part of the Wi-Fi. The other part is HiperLAN family. Wi-Fi can provide coverage of the radio of the hundred meters at most. The initial throughput of the Wi-Fi is just 11Mbps, but with the development, its current throughput is already increased to 54Mbps. With some Wi-Fi devices, the throughput can be raised to 108Mbps nowadays. Furthermore, the studies show that the throughput of the Wi-Fi can go up to 540Mbps in the future. The most popular networks that are suitable to adopt the VoIP are the famous Wi-fi and recently Wimax.[10]

As a new member of the wireless network family, Wimax draws more attention because of its better performance as it can provide high quality service, larger coverage and super security.

The figure 2 shows the interface that we developed over the campus. To use the software, it needs the authorization of sever. Once the users register on the sever, they will get a number as his telephone number to be used for communication over the sever. The users can fulfill the blankets to connect to the sever. We provide a simple and friendly interface to all the users. On the right of the interface, there is a section to

display the information on the users calling, such as whether connected or if the line is occupied or free to use.

Figure 2. Main interface of the software.

4 Experimental results

We had measured the wireless network over the campus to investigate its performance for the VoIP technology.

First of all, with the Q check, a small tool to verify the response time, throughput, streaming and traceroute for different kinds of the protocol like TCP, UDP and so on. Figure 3 is the result of the response time in Q check with the protocol UDP.

Figure 3. Response time with Q check

Netquality is a software that uses the test clients to do pseudo calls and records the delay, jitter, losts and max.gap. It firstly research for the ordered range of the IP address to find out the available ones to do pseudo calls. The figure shows the differences parameters like loss, jitter and delay. Results of the Netquality are shown in the figure 4.

From the figure 5, it's easy to find out when and where the jitters take place and to examine the exact information of the jitters.

The figure 6 shows the usage of the wireless network when there are only one Skype call that goes on the network.

At last, we use the «wireshark» to explore the protocol SIP, such as how does it works to start and end the conversation and the content in the SIP package. Figure 7 and 8 are a pair of the graph to show the results about SIP.

NetQuality Profile Results - Teststart: 28.03.2008 16:08:25														
Customername and Info: ETUD														
Profilename: G_711 (150ms)-10secs														
Sequence finished: 2 Test(s) successful, 5 failed. Failures reasons: Loss: 4 Jitter: 2														
Table View Text View														
Client Address	Test-Type	#Simul. Conn.	Start Time	End Time	Result	Loss %	Pseudo Loss %	Avg Delay	Max Delay	Avg Jitter	Max Jitter	Max Gap (msecs)	Broadc. Avg./sec	%-Loss ***
139.103.128.126	G.711 (150ms)	1	28.03.2008 16:08:25	28.03.20	Completed.	2,2%	2,2%	34	93	26	59	150	5	0,0%
139.103.128.202	G.711 (150ms)	1	28.03.2008 16:08:36	28.03.20	Completed.	0,0%	10,2%	106	510	66	404	0	5	0,0%
139.103.129.40	G.711 (150ms)	1	28.03.2008 16:08:46	28.03.20	Completed.	0,0%	0,0%	22	102	14	80	0	5	0,0%
139.103.129.63	G.711 (150ms)	1	28.03.2008 16:08:56	28.03.20	Completed.	1,6%	6,3%	159	3098	134	2939	150	4	0,0%
MAC001EC2116A5	G.711 (150ms)	1	28.03.2008 16:09:07	28.03.20	Completed.	0,0%	0,0%	66	179	45	113	0	7	0,0%
PAVILLON139.103.103	G.711 (150ms)	1	28.03.2008 16:09:27	28.03.20	Completed.	0,0%	9,7%	107	286	42	179	0	5	0,0%
PHILIPPE.PC(139.103.103.103)	G.711 (150ms)	1	28.03.2008 16:09:46	28.03.20	Completed.	0,0%	2,1%	109	2898	118	2789	0	3	0,0%
7/7 Tests	---	---	---	---	---	0,5 %	4,4 %	86,1	1023,	63,6	937,6	42,9	4,9	0,0

Figure 4. Results of the Netquality pseudo calls with the clients

Figure 5. Result for jitter

Figure 6. The Use of the wireless network

776	26.011696	198.65.166.131	192.168.2.2	SIP	Request: BYE sip:diegosaura@139.103.68.34
777	26.387190	192.168.2.2	198.65.166.131	SIP	Status: 200 OK
778	26.388562	198.65.166.131	192.168.2.2	SIP	Request: BYE sip:diegosaura@139.103.68.34
779	26.390931	192.168.2.2	198.65.166.131	SIP	Status: 200 OK
780	26.540008	198.65.166.131	192.168.2.2	SIP	Status: 200 OK
781	26.570978	198.65.166.131	192.168.2.2	SIP	Status: 200 OK
782	27.181563	192.168.2.2	198.65.166.131	UDP	Source port: sip Destination port: sip

Figure 7 SIP starts a conversation

2	1.727202	198.65.166.131	192.168.2.2	SIP	Request: BYE sip:diegosaura@139.103.68.34
3	1.728378	192.168.2.2	198.65.166.131	SIP	Status: 200 OK
4	1.848607	198.65.166.131	192.168.2.2	SIP	Status: 200 OK
5	1.995034	BelkinIn_02:e3:0f	Spanning-tree-(for-br	STP	Conf. Root = 32768/00:1c:df:02:e3:0f Cost = 0 Port
6	2.616921	192.168.2.2	198.65.166.131	UDP	Source port: sip Destination port: sip
7	3.990088	BelkinIn_02:e3:0f	Spanning-tree-(for-br	STP	Conf. Root = 32768/00:1c:df:02:e3:0f Cost = 0 Port
8	4.736721	198.65.166.131	192.168.2.2	SIP/SDP	Request: INVITE sip:diegosaura@139.103.68.34:5060, w
9	4.738508	192.168.2.2	198.65.166.131	SIP	Status: 100 Trying
10	4.738812	192.168.2.2	198.65.166.131	SIP	Status: 180 Ringing
11	5.985128	BelkinIn_02:e3:0f	Spanning-tree-(for-br	STP	Conf. Root = 32768/00:1c:df:02:e3:0f Cost = 0 Port
12	7.299720	192.168.2.2	198.65.166.131	SIP/SDP	Status: 200 OK, with session description
13	7.563064	198.65.166.131	192.168.2.2	SIP	Request: ACK sip:diegosaura@139.103.68.34

Figure 8. SIP ends the conversation

5 Conclusion

Nowadays, the wireless network becomes the first choice of most university because of its convenience and high speed which is equivalent as that of the ethernet. At the sametime that we can talk with each other without the traditional telephony, we also need a better communication tool. We have designed a wireless solution and presented in this paper some of the measurement results. The developed application can be well adopted by the wireless network environment of the campus.

Reference

- [1] Xudong Wang, Abhishek Patil, Weilin Wang, *VoIP Over Wireless Mesh Networks: Challenges and Approaches*, WICON'06, August 2-5, 2006, Boston, MA, United States
- [2] Yukuan Ma, *The design and implement of VoIP system*, The thesis of Jilin University, 2005.9
- [3] Dragos, Niculescu, Samrat Ganguly, Kyungtae Kim, Rauf Izmailov, *Performance of VoIP in a 802.11 Wireless Mesh Network*,
- [4] Jen-Jee Chen, Ling Lee, Yu-Chee Tseng, *Integrating SIP and IEEE 802.11e to Support Handoff and Multi-grade QoS for VoIP Applications*, Q2SWinet'06, October 2, 2006, Malaga, Spain
- [5] Tariq Latif, Kranthi Kumar Malkajiri, *Adoption of VoIP*, 2007.03
- [6] Vijay Sundar Rajaram, *Session Initiation Protocol for Wireless Channels*, 2006
- [7] Ahmad M. Kholaiifa, Terence D. Todd, *WLAN VoIP capacity allocation using an adaptive voice packetization server*
- [8] Mirosław Narbutt and Mark Davis, *Gauging VoIP call quality from 802.11 WLAN resource usage*, WoWMoM'06
- [9] Nidhi Hegde, *Evolution of WLANs towards VoIP traffic and higher throughputs*, AccessNets'06, September 4-6, 2006, Athens, Greece.
- [10] Olli Alanen, *Multicast Polling and Efficient VoIP Connections in IEEE802.16 Networks*, MSWiM'07.