Voice over Wireless Data Networks

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Abstract

As wireless technology becomes increasingly available it only makes sense that applications like Voice over IP (VoIP) would be implemented. The merging of two areas, VoIP and wireless networks, has created Voice over WiFI (VoWiFi). Problems with VoIP are not only ported to wireless networks, they are also magnified. Mobility is an issue with wired networks, however wireless networks require mobility on a much larger scale. This paper focuses on some underlying issues of VoIP and VoWiFi. It also presets some background on protocols and codecs used with telephony data.

Keywords

VoIP, VoWiFi, 802.11i, 802.11r, 802.11e, Codecs, RVP/IP, SIP

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Introduction

Voice over IP (VoIP) is an area of research and development that has been reported on extensively. This area as created to leverage the current wired computer networks to transmit multimedia. In particular, the transmission of voice has been a very important area. Wired networks can provide all the necessary functions to transmit voice traffic. However, there are inherent problems with voice over wired IP networks. An example of such a problem can be seen with the TCP/IP protocol. This protocol

does not have any quality of service guarantees. Such guarantees are important with multimedia. With voice a caller and the called would prefer little to no delay.

As Wireless technology becomes more popular and implemented in mainstream networks, it is only natural for applications such as VoIP to be implemented. The new joint area is called Voice over Wireless Data Networks. This area covers Voice over Wi-Fi (VoWiFi) as well as Voice over Wimax, but excludes cellular networks. The idea of VoWiFi is to utilize existing wireless technology that is used for computer networks to transport voice traffic. Such computer networks can be thought of as Data Networks, since the original purpose was to transmit data from computer to computer.

This newly merged area of research maintains most, if not all, of the wired network problems with voice traffic. In addition, mobility and security become a bigger problem. With individuals traveling to and from subnetworks issues of handoff and connectivity become apparent. This report will identify three problem areas and discuss solutions that are currently under development. It will also present some fundamental information about codec, and protocols that have been developed for voice traffic.

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2. Main Issues and Solutions

Voice over Wireless Networks has the same issues as VoIP. Quality of Voice, and Scalability are important to the use of VoIP. In addition to these issues, Quality of Service, Mobility, and Security of telephony traffic are more problematic in wireless networks. These issues or problems are currently being explored for wired and wireless networks in general. These solutions are being adopted within implementations of VoIP and VoWiFi products. As more products are created for telephony over data networks, more solutions are added by creating new protocols and services that help maintain voice traffic.

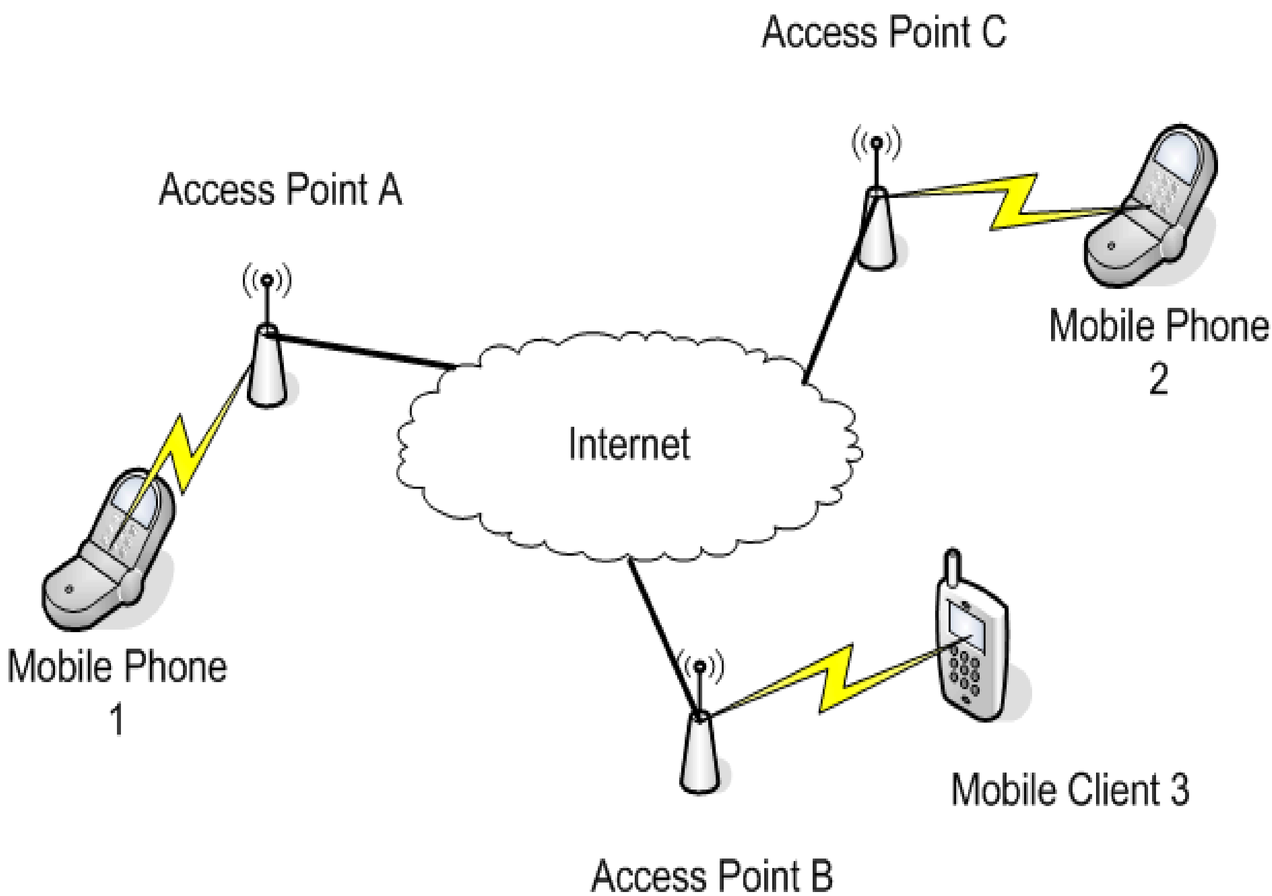

Fig 1. Example Voice over WiFi network

2.1 Quality of Service

The current group that is adding Quality of Service (QoS) to the wireless standards is 802.11e. This standard includes two types of QoS: Prioritized, and Parameterized. Prioritized QoS implements different queues for priorities. Traffic is given a tag that is used to target a queue. By using tagging and different queues types of traffic (i.e. voice) can get priority. This implementation allows for priority, however there is no guarantee on the amount of bandwidth given to a priority or queue. The second type of QoS, Parameterized QoS, allows for a bandwidth guarantee. This guarantee (for streams) provides applications such as VoWiFi the ability to maintain throughput. [VoWiFi Standards]

2.2 Mobility

When talking about wireless networks mobility is an important consideration. Applications running on a network need the ability to continue running when transferring from one area to another. For example, in Figure 1 if Mobile Phone 2 where to move further south it would require it to switch to Access Point B. In this example Mobile Phone 2 would need to negotiate with Access Point B and prepare for the switch. After switching Mobile Phone be would have to deassociate with Access Point C. In addition to connecting to one access point it also need to be able to disassociate from it current access point. The 802.11r standard is currently being developed that allows for fast hand-offs. The standard allows a user to connect and negotiate security and QoS settings before dissociating with it's current access point. This allows a mobile user to maintain service as access points are changed. For indepth information refer to the IEEE 802.11r standard. It is also explained at [802.11r explained].

2.3 Security

Security is an important topic in networking. It is only natural that security concerns are present in both wireless networks and the applications running on the networks. With in the application of Voice over Wireless networks the main security concerns revolve around confidentiality, data integrity, and data source authentication. The wireless standard 802.11i addresses these problems within wireless networks and provides the necessary functionality. There are currently two versions for security: Temporal Key Integrity Protocol, and Counter Mode with Cipher Block Chaining Message Authentication Code Protocol.

Temporal Key Integrity Protocol (TKIP) is an important first step in security. The TKIP algorithm is designed so that it can be implemented in current hardware with only firmware upgrades. It uses an algorithm called Michael to check the integrity of the message. Michael uses the RC4 hashing algorithm, however it is used in a secure form. In addition TKIP allows for rapid re-keying.

A more secure protocol is called Counter Mode with Cipher Block Chaining Message Authentication Code Protocol (CCMP). CCMP is much stronger than TKIP. It utilizes the Advanced Encryption Standard (AES). This protocol requires new hardware that can implement the AES encryption and decryption. [VoWiFi Standards]

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3. Codecs

The quality of voice is a characteristic on digital telephony. Since VoIP and VoWiFi combine digital telephony and networking technologies they also have the quality of voice characteristic. An important element that controls the quality of voice is the compression, and conversion of analog to digital (codec) used on the voice traffic. There are numerous codecs defined for use with voice traffic. The following sections will introduce and give an overview of a few current codecs in use.

3.1 Global System for Mobile communications

The Global System for Mobile communications (GSM) is a popular codec used in cell phone networks in Europe. The codec samples the audio signal to produce a digital equivalent. The current sampling is determined by the previous samples. This assumes that the streaming data (voice in this case) does not change quickly. The samples are then passed to the codec at 13 kbps. The sampling rate is 8 kHz and the frame size is 22.5 ms. A more detailed description of the GSM protocol can be found at [GSM].

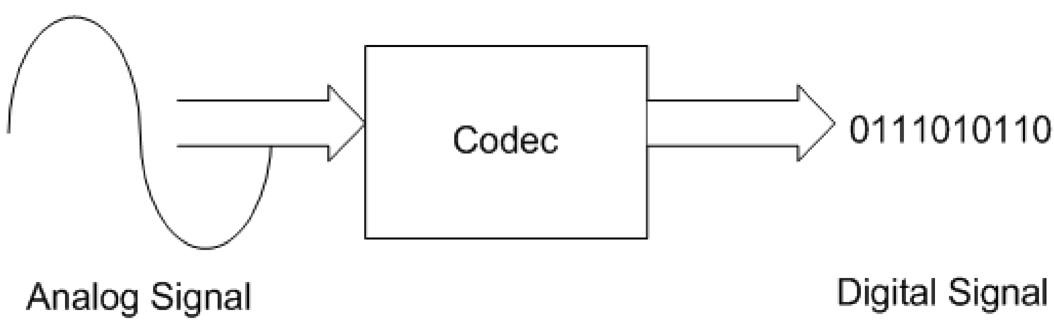

Fig 2. Analog Voice is Sampled and may be compressed by the Codec into a Digital Form

3.2 G.711

The G.711 codec is a standard adopted in 1988, that is used in modern digital telephony. The advantage of this codec is that there is no compression of the signal. This means that there is no additional processing needed for compression and decompression. Using an uncompressed signal means that there is an increased amount of bandwidth required for transmission. This codec can require up to 84 Kbps when all the TCP/IP headers are taken into consideration. This codec has a sampling rate of 8 KHz, and a bit rate of 32 kb/s. Pulse code modulation (PCM) is used to sample the stream. There are two variations of this codec: U-law and A-law. For more information on G.711 please refer to [G.711].

3.3 G.722

This codec divides a 16 kHz frequency band into two sub-bands. Adaptive Differential Pulse Code Modulation (ADPCM) is used on each of the sub-bands for coding. This codec provides a bit rate of 64 kb/s with a sampling rate of 16 kHz. More information of the codec can be found at [G.722].

3.4 G.723.1

The G.723.1 codec is used for compressing audio at very low bit rates. The principle application that this codec is targeting is visual telephony. It is part of a larger family of standards, H.324. The codec has a dual rate speech coder that can handle a data rate of 5.6 kb/s and 6.3 kb/s. The codec uses a sampling rate of 8 kHz and a frame size of 30 ms. A more in depth look at the codec can be found at [G.723.1]. In order to use this codec a license must be obtained from Sipro Lab Telecom.

3.5 Codec Summary

One of the most common ways to determine the quality of a codec is to determine the Mean Opinion Score (MOS). The MOS is a value between one and five. A score of one means there is a communications breakdown. An excellent score is a five. This is equivalent to a perfect AM radio reception. Current land line telephony achieves a score of four. The closer a codec is to a score of four represents a small difference between the codec and current land line systems. The following table was compiled primarily from [Codec Table], but also information found in numerous journal papers.

Codec	Bitrate (kb/s)	Sampling (kHz>	Frame Size (ms)	MOS
GSM	13	8	22.5	3.5
G.711	32	8	Sample	4
G.722	64	16	Sample	3.6
G.723.1	5.6/6.3	8	30	3.8
G.729	8	8	10	4.02

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4. Protocols

There have been a few protocols that have been developed for voice and multimedia data. The protocols help to define what is needed in a physical network to help facilitate voice traffic. The first protocol that was developed that identified and solved many issues of voice traffic was called H.323. After discussing the Session Initiation Protocol will be introduced and described. Within this paper

these are the only two that will be discussed.

4.1 H.323

H.323 is the International Telecommunication Unit's (ITU) protocol for establishing VoIP connections. The standard was the first to solve the issues of VoIP over networks. The standard contains three primary components: Call Processing Servers, Media Gateways, and Gatekeepers.

Call processing servers handle call routing. In addition, it allows for communication to VoIP gateways and any end user devices. The media gateways provide an interface with non-H.323 networks in addition to being the protocols termination node. The gatekeepers (although not necessary) provide the functions of call admission control, call signaling and bandwidth management as a co-located unit. The gatekeepers allow the protocol to be highly scalable by taking the call control and management from the gateways. For a more look at the protocol refer to [H.323].

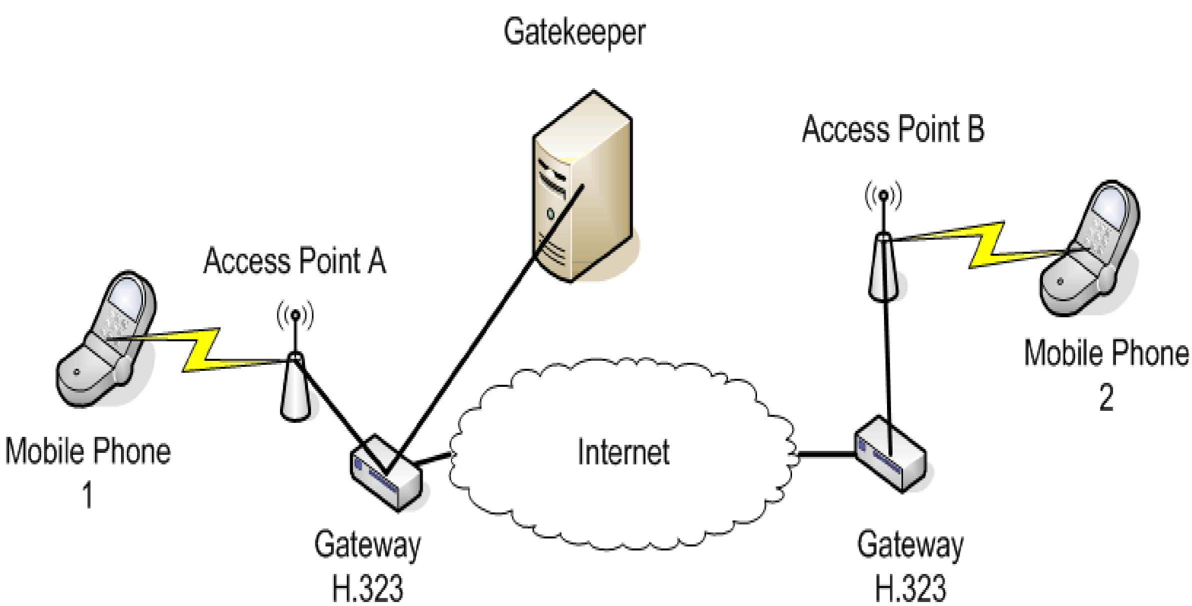

Fig 3. Example of a H.323 network with a Gatekeeper

Within Figure 3 it can be seen how an example voice over wireless networks would incorporate the H.323 protocol. If Mobile Phone 1 were to place a call it would contact the gatekeeper for the address of the second party (Mobile Phone 2). It would then contact Mobile Phone 2 and they would use the H.323 protocol to set up a connection and start data transmission.

4.2 Session Initiation Protocol

An application-layer control protocol called Session Initiation Protocol (SIP) is utilized in VoIP to establish, modify and terminate connections. The protocol allows for call forwarding, authentication, multicast conferencing, and automatic call distribution. SIP allows for mobility by providing a transparent name mapping. There are six methods that are supported in the SIP protocol: INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER. The details of the SIP protocol can be found in the RFC2543 [RFC2543].

Over the past few years the original SIP protocol has been extended and modified to alleviate problems. Currently there are three additional RFCs that extend the protocol. RFC4320 [RFC4320]is provides a solution to the SIP non-INVITE operation. This RFC shows modifications that can reduce network traffic and improves the robustness of SIP networks. Extending the reason header for preemption

events is described in RFC4411 [RFC4411].

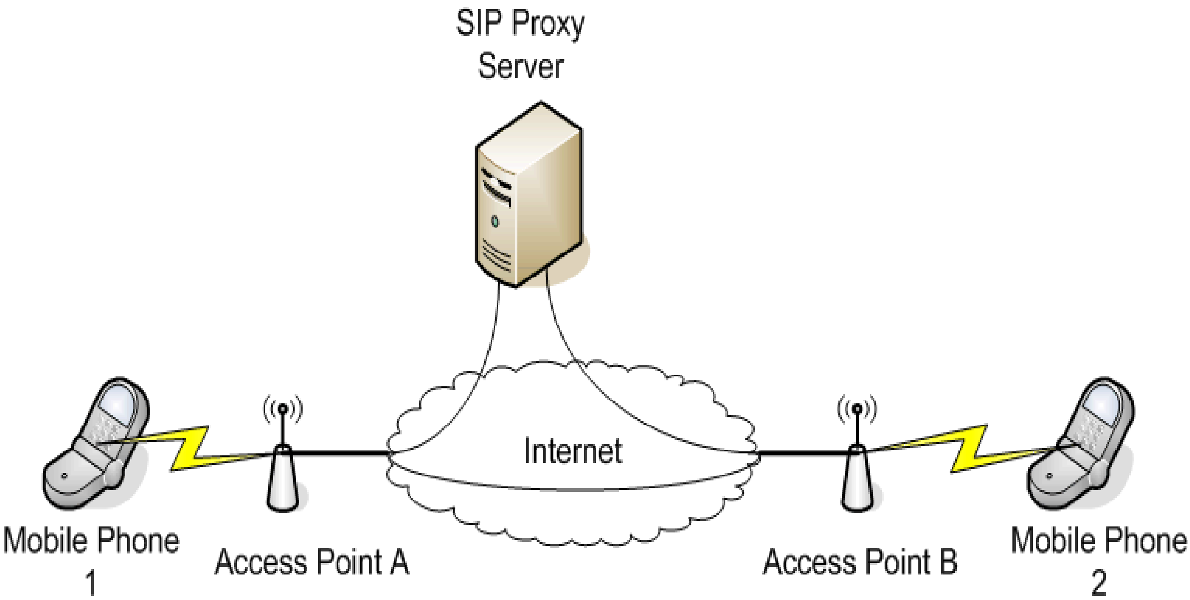

Fig 4. Example of SIP being used for Voice over wireless network

Figure 4 shows an example of the protocol SIP being utilized for voice over a wireless network. When working in a redirect mode the connection sequence would be as follows. Both phones would register with the SIP proxy server. Mobile Phone 1 would they send a request to the server trying to connect to Mobile Phone 2. The SIP server would then send Mobile Phone 2 a message stating that there is a connection request. When Mobile Phone 2 picks up the SIP protocol then sets up an RTP connection between the two phones. [Juniper].

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5. Summary

Voice over wireless data networks suffer from the same problems as Voice over IP. Some of these problems are magnified due to the natural mobility wireless products offer. With in the 802.11 standards there are standards that are being developed to provide solutions to some of these issues. For security, 802.11i is under production. Mobility should be increased with the hand-off characteristics being developed in 802.11r.

Quality of Voice is a complex issue and is being addressed in at least two ways. In the standard 802.11e priority queues are utilized to help ensure the throughput of Voice and other multimedia traffic. Voice quality is also determined by the codec that is chosen and implemented. The codec is important, because the sampling and the compression ratio can drastically decrease the quality. A score called MOS is used to determine the quality of telephony.

Two protocols were also introduced and discussed. These protocols were created with voice and multimedia in mind. They are used to help provide a basis for creating, maintaining, and closing connections between callers. The first protocol introduced was the H.323, but it is no in competition with the protocol called SIP.

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[RFC4412] "Communications Resource Priority for the Session Initiation Protocol (SIP)", February 2006, 36 pages, ftp://ftp.rfc-editor.org/in-notes/rfc4412.txt
Defines resource priority for SIP.

[Juniper] "Voice Over IP 101 Understanding VoIP Networks", January 2006, 23 pages,
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Describes the various protocols and gives examples.

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List of Acronyms

IP	Internet Protocol
VoIP	Voice over IP
VoWiFi	Voice over WiFi
GSM	Global System for Mobile communications
MOS	Mean Opinion Score
TKIP	Temporal Key Integrity Protocol
CCMP	Cipher Clock Chaining Message Authentication Code Protocol
AES	Advanced Encryption Standard
PCM	Pulse-code Modulation
ADPCM	Adaptive Differential Pulse Code Modulation
SIP	Session Initiation Protocol

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