Analysis of VoIP traffic over WiMAX

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Abstract: *Voice over Internet Protocol (VoIP) is a rapidly emerging technology for voice communication that uses the ubiquity of IP-based networks to deploy VoIP-enabled devices in enterprise and home environments. VoIP-enabled devices-such as desktop and mobile IP phones and VoIP gateways decrease the cost of voice and data communication, enhance existing features, and add compelling new communication features and data services. VoIP applications (Skype, Google Talk, and MSN Messenger) are being widely used in today's networks challenging their capabilities to provide a good quality of experience level to the users. In particular, new wireless broadband technologies, such as WiMAX. In this paper the authors have discussed about the issues and challenges when voice goes over wireless.*

Keywords: VoIP, WiMAX, Latency, Jitter, MOS.

1. Introduction

Broadband wireless access (BWA) technical solutions and products have been available for some time. These technologies have been primarily focused on providing high data rate connectivity wirelessly between fixed stationary sites. Many examples of these types of applications include building to- building bridging and providing high-rate connectivity to remote sites, such as broadcast towers, where the installation of wired infrastructure is not possible. The IEEE 802.16 BWA technology often referred to as worldwide interoperability for microwave access (WiMAX) or Wireless MAN, is intended to provide a standardized BWA solution to provide "broadband wireless to the masses'' and is so anticipated that it has even been characterized by some as a threat to the long-term viability of several existing wireless technologies (including IEEE 802.11-based wireless local area network [WLAN] technology, broadband residential Internet technologies such as digital subscriber line [DSL] and cable), even viewed by some as a competitor to third-generation (3G) cellular technologies.

WiMAX is a standards-based technology, which enabling the delivery of last mile wireless broadband access as an alternative to wired broadband like cable and DSL. WiMAX provides fixed, nomadic, and portable and soon, mobile wireless broadband connectivity without the need for direct line-of-sight with a base station. In a typical cell radius deployment of three to ten kilometers, WiMAX systems can be expected to deliver capacity of up to 40 Mbps per channel, for fixed and portable access applications. This is enough bandwidth to simultaneously support hundreds of businesses with T-1 speed connectivity and thousands of residences with DSL speed connectivity. Mobile network deployments are expected to provide up to 15 Mbps of capacity within a typical cell radius deployment of up to three kilometers.
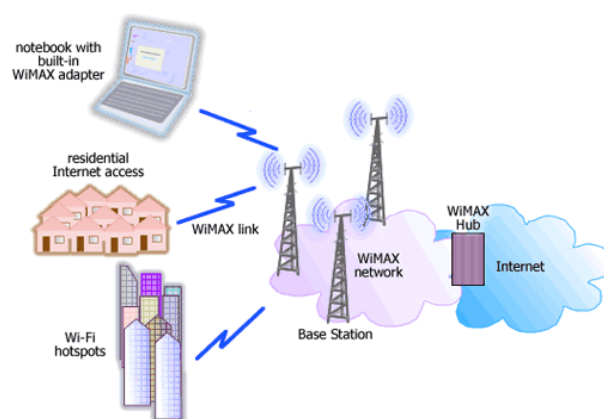

Figure: How WiMAX works

WiMAX is the next generation evolution in wireless technology and enables high-speed connectivity to meet the increasing demand for broadband Internet at home, in the office, or while on the go. The IEEE 802.16e-2005 standard for portable devices enables a new era of high throughput and high delivered bandwidth together with exceptional spectral efficiency. WiMAX based on advanced technologies such as OFDMA and MIMO. The rest of the paper is organized as follows: section II presents the VoIP principle, VoIP technology and VoIP architecture, in section III consists issues and challenges of VoIP over wireless, the conclusion along with future scope are discussed in section IV.

2. VoIP Concepts

VoIP stands for Voice over IP, or in other words, telephone service over the Internet. VoIP is simply the transmission of voice conversations over IP-based networks. Although IP was originally designed for data networking, its success has led to its adaptation to voice networking. Today, VoIP has begun to be accepted by more and more consumers and business users. There are three major areas of VoIP. First, VoIP quality measurement; second, methods to improve VoIP quality; third, VoIP protocol and traffic analysis which aids in the understanding and design of VoIP systems, from which more useful predictive models can be generated.

As according to the need of modern telecommunication system Wireless networking has become an essential part the demand of high speed data transfer with high quality is being the leading factor for the evolution of technologies like WiMAX and WLAN and is still increasing day by day. Therefore, new ways to improve quality and speed of connectivity are being searched for. Moving towards the fourth generation communication networks, integrated networks are coming into operation. In same manner voice over IP is expected to be a low cost communication medium. The voice codecs are big constraints which influence the quality of the voice in a high data rate communication network. Therefore, before real time deployment of VoIP over a network it is essential to evaluate the voice performance over altering networks for various codecs.

VoIP have been widely accepted for its cost effectiveness and easy implementation. A Voice over internet protocol (VoIP) system is divided into three indispensable components, namely 1) codec, 2) packetizer, and 3) play out buffer. Analog voice signals which are to be transmitted compressed, and encoded into digital voice streams by the help of codecs. The output digital voice streams are then packed into constant-bit-rate (CBR) voice packets with the help of the packetizer. A two way conversation is very sensitive to packet delay jitter but could tolerate certain degree of packet loss. Hence a playout buffer must be used at the receiver end to smooth the speech by eliminating the delay jitter. Quality of noise sensitive VoIP is generally measured in terms of jitter, MOS and packet end-to-end delay. Perceived voice with zero jitter, high MOS and low packet end-to-end delay is assumed to be the best. Before transmitting voice over internet which is an analog signal should be converted into digital format. To obtain digital format of the analog signal process is utilized which is called encoding and converse is called decoding and both are performed by voice codecs. As bandwidth is enormous concern, compression techniques are utilized to reduce bandwidth consumption. But problem related by using codecs is the overhead of algorithmic delay, thus codec is assumed to provide good quality even after compression, with minimum delay.

- **How It Works:** The encoder is the packetized, which encapsulates a certain number of speech samples into packets and adds the RTP, UDP, IP, and Ethernet headers. The voice packets travel through the data network to the receiver where an important component called the playback buffer is placed for the purpose of absorbing variations or jitter in delay and for providing a smooth play out. Packets are then delivered to the de-packetized and eventually to the decoder, which reconstructs the original voice.
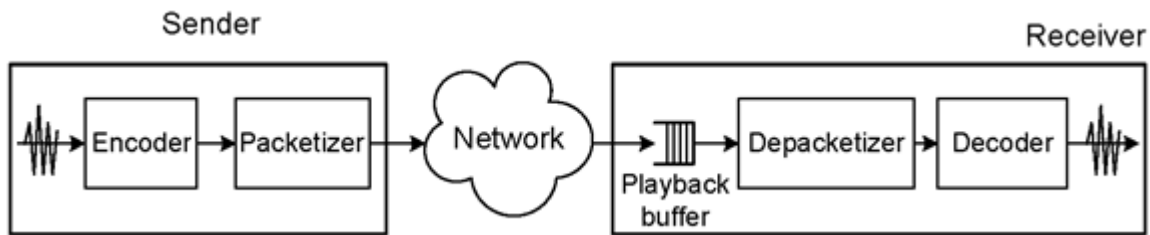

Figure 1: VoIP Technology

- **VOIP Architecture:** At the sending end, the original voice signal is sampled and encoded to a constant bit rate digital stream. The digital stream can then be easily compressed. This digitized and compressed data is then encapsulated into packets of equal sizes for easy transmission over the Internet. Along with the compressed voice data, these packets contain information about the packet's origin, the intended destination, and a timestamp that allows the packet stream to be reconstructed in the correct order. These packets flow over a general-purpose packet-switched network, instead of traditional dedicated, circuit-switched voice transmission lines. At the receiving end, the continuous stream of packets are de-packetized and converted back into the analog signal so that it can be detected by the human ear. In general, this means voice information is sent in digital form in discrete packets rather than using the traditional circuit-committed protocols of the Public Switched Telephone Network (PSTN). In addition to IP, VoIP uses the Real-Time Transport Protocol (RTP) to help ensure that packets get delivered in a timely way. Over the last few years, VoIP has become increasingly popular and is already starting to replace existing telephone networks.
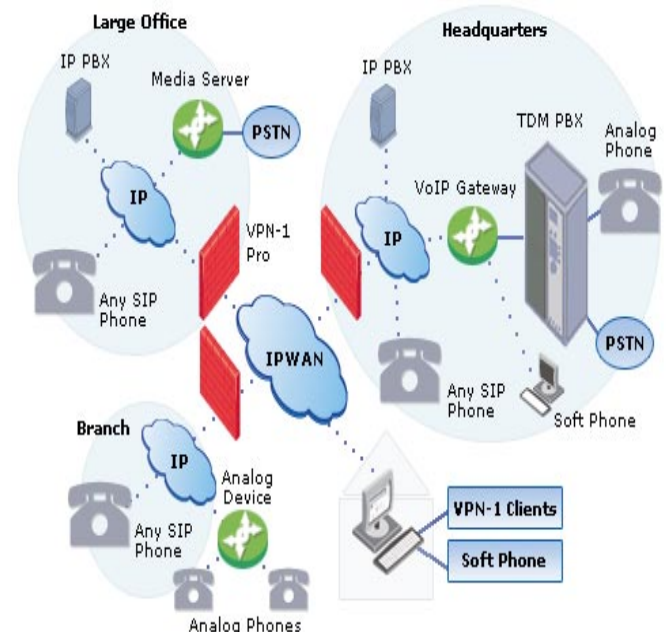

Figure 2: VoIP Architecture

3. Issues and Challenges of VoIP over Wireless (WiMAX)

- **Mean Opinion Score (MOS):** As voice communication is noise sensitive. Noise is the main cause due to which the signal to reach the destination with a lead or lag in the time period. The deviation in the signal characteristics is called jitter. Lead causes negative jitter and lag causes positive jitter and both will degrade the voice quality. The time taken by voice to be transmitted from the source to the destination is called packet end-to-end delay. This delay should be very less for voice communication. Perceived voice quality is classically estimated by the subjective mean opinion score (MOS), an arithmetic average of opinion score. MOS of a particular codec is the standard mark given by a panel of auditors listening to various recorded samples. This will range from 1(unacceptable) to 5 (excellent). It will depend on delay and packet lost by the network.

Table 1: Mean Opinion Score

Quality Scale	SCORE	Listening effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with reasonable effort

- **Jitter:** The variation in arrival time of consecutive packets is called jitter. Before decoding, packets arrive to limited size buffer however some packets may lost or arrive out of order. Jitter can be calculated by computing the difference delay of packets over a period of time.
- **Packet end-to-end delay:** The end-to-end delay can be measured by calculating the delay from the speaker to the receiver. This includes network delay, encoding and decoding delay, and compression and decompression delay.
- Packet loss does excessive damage to the voice signal, as retransmission cannot be considered as an option while transmitting voice. Loss of voiced frames at unvoiced/voiced transition causes significant degradation of the signal. Advanced error detection and correction algorithms are used to fill the gaps created by the dropped packets. A sample of the speaker's voice is stored and is used to create a new sound from an algorithm which tries to approximate the contents of the dropped packets or lost packets.
- **Latency:** Latency is the time taken for a packet to arrive at its destination. Latency also happens due to Packet switching overhead and Congestion. Latency may result in voice synchronization problems.
- **Bandwidth:** When bandwidth is shared between voice and computer data, certain bandwidth may have to be allocated for voice communication on a network. So bandwidth allocation is also the major issue in VoIP.
- **Security:** Securing the voice communication is also a big challenge for VoIP over WiMAX as care has to be taken that it cannot be eavesdropped or intercepted. The Double encryption process - X.509 for Authentication and 152-bit AES, 3DES or 56-bit DES for data flow ensure the transmission is secure and eavesdropping is very difficult on the traffic.
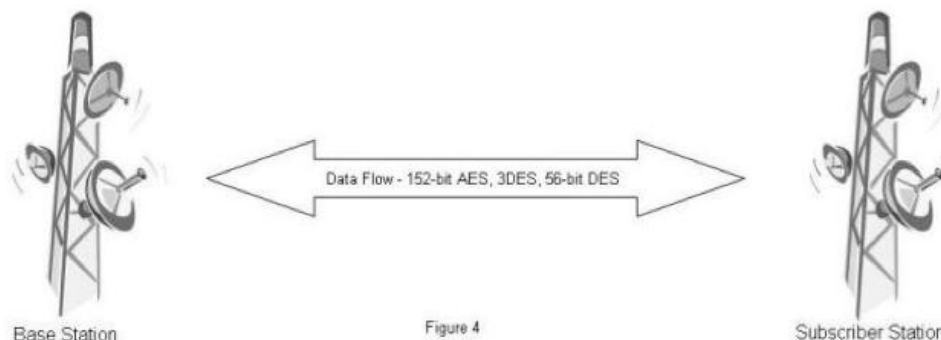

Figure 3

Figure 4

When a Subscriber Station (SS) needs to associate with a Base Station (BS), it sends an authorization request along with authentication information in a X.509 certificate (Figure 3).

The BS after verifying the certificate responds by sending the authorization message which has the authorization key encrypted with subscriber's public key, to enable the subscriber to register with the network. An IP address is given to the SS by the DHCP. The DHCP server also provides the address of the TFTP server, from where the SS gets the vendor specific configuration information file.

After this, the BS accepts the subscriber. The data stream is further encrypted using 56-bit DES, 3DES or 152-bit AES (Figure 4). This prevents the possibility of eavesdropping on the data and theft of service as the links between the BS and SS are encrypted. Also WiMAX has built in virtual LAN (VLAN) support which provides protection for data transmitted by multiple users on the same BS.

4. Conclusion

It has been concluded that WiMAX can not only be used to fulfill the demand for high internet speed but can also be used to provide voiceover- IP services. The low-latency design WiMAX makes it possible to deliver VoIP services more effectively and VoIP technologies may also be used to provide innovative services like voice chatting, push-to-talk and multimedia chatting. Several good works have discussed the capacity and performance of WiMAX network. But they appear a scope for a comparative discussion of the performance of a WiMAX network with respect to the application of VoIP.

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660