# Enhanced Quality Management of VoIP Service coexistence over WAN & WiMax Architectures for Streaming Video in Mobile Appliance

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## ABSTRACT

This research article describes about the voice call transmission among WiMax coverage area. The theme of this work is to obtain better voice clarity than mobile communication networks. The objective of fourth generation networks is to provide larger coverage, low power handling of base stations and good compatibility. Similarly, this work examines the best voice clarity with reliable quality of network management in the Long Team Evaluation (LTE) streaming coexisting standards. It offers comparative analysis of the two scenarios. They are the different coverage locations (50x50 km and 100 X 100 km), a large number of mobile users, same configuration of connection setup with devices and the diverse interference among users. Finally the high coverage area has achieved the results better than lower coverage region.

**Keywords :** **Streaming, VoIP, WiMax, Network management, Mean Opinion score, G.711 voice CODEC**

## 1. INTRODUCTION

WiMAX is one of the wireless WAN (Wide Area Network) technologies. The WAN is integrating the WiMax and VoIP over services. VoIP is a rapidly growing mobile internet IP based technology. It provides the cheap call setup connection and better coverage than other location based networking technologies. The drawbacks of TDMA technology are i) Multipath distortion ii) every user has a predefined in the time slot iii) reestablishment connection setup. The above drawbacks are rectified by VoIP over IP based data communication networks. Upcoming 4$^{th}$ generation network objectives are, to reduce the complexity, power/energy dissipation levels and improved coverage area. This article suggests that WiMAX could be a perfect proposal for an internet protocol (IP) based on video services. The PSTN is one of the telephone networks which achieve good voice quality (MOS is more than 4) in the second generation.

Nowadays, the world goes behind wireless technology built the quality of voice is very poor due to more packet loss and jitter. In this paper, WiMax scenario is obtained achieving good voice quality which is nearly equal to the voice quality of PSTN/Mobile Networks. The lots of factors are affected in the mobile networks such as network maintenance in the larger coverage area, utilization of bandwidth, installation cost, subscriber billing charges, interferences among mobile users and propagation loss. The above drawbacks are overcome easily by VoIP technology. The VoIP transmission provides high quality of voice calling/called signaling, secure communication through IP devices. The integration of VoIP technology and WiMAX makes a cheap and free voice calls (gtalk, Skype) using the notebook, laptop and phone. While transmitting voice calls in wireless medium, the parameters like high speed delivery, low jitter, reduced end-to-end delay, improving throughput, utilization by users and Mean Opinion Score value are focused mainly. WiMAX based IP technology support high speed internet access, telephone services, voice application, video conferences, video streaming services, etc. VoIP is one of the signaling and speech transmission protocol. In VoIP service, the voice is converted into data packets which is done by voice CODEC encoder and are transmitted to the destination through Internet Protocol (IP). In VoIP is used in one of the best voices CODEC (encoder scheme) is G.711. The some other standard protocols in VoIP are H.323, SIP (Session Initiation Protocol), MGCP (Media Gateway Control Protocol), RTP (Real-time Transport Protocol), SDP (Session Description Protocol) and IAX (Inter-Asterisk exchange).

The LAN and WAN network architectures are supported by VoIP services. The Investigation of VoIP testing and installation has been done by LAN and WAN networks to help of gateways, servers and router devices (Cisco, IBM, HP, DELL, Mainframe). The performance analysis of VoIP over services is based on this LAN network that has following considerations, i) speed of delivery ii) the voice quality iii) short delay and iv) jitter and also other LAN architectures (Ethernet, token bus, token ring , FDDI (fibre distributed data interface)) are supported with VoIP services. The Ethernet architecture is one of the popular standards particularly, used as video LAN with wired IP based applications. The simulation parameters are evaluated by following results, i) no short delay, 2) jitter (delay variation), 3) only affected by the collision of packets in topology structure, 4) few errors appear in Ethernet VoIP services. Hence LAN is the recommended standard for VoIP service in order to improve the Qos with security Issues. The WAN gives better VoIP services than a LAN. The major issues focused in WAN networks are: i) bandwidth is limited ii) end to end delay is longer than LAN networks iii) jitter iv) transmission rate is low but VoIP signalling protocol is enough transmission rate so it doesn't require the retransmission in WAN networks. While receiver IP devices are compensating the jitter parameters, the arrival of incoming packets is converted into an analog sound. The receiver IP is simulated the VoIP packets in order to eliminate the holes in between words when the packets are lost during transmission. Finally, such as voice packets are modified by CODEC-compressor scheme. In this simulation time reduces the jitter, packet loss, extra delay occurred between IP devices.