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LOCALIZED WIFI ENABLED VOICE COMMUNICATION NETWORK FOR EDO STATE POLYTECHNIC

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ABSTRACT

The rapid growth that has characterized the internet in the last few years has resulted in the emergence of several related and dependent technologies. Among these technologies is the internet protocol-based telephone system. This telephone technology has received wide acceptance and usage as a result of the many advantages it has over the traditional telecommunication system which among other things is the provision of an economical way to make international communication. This technology involves a phone system that uses an internet connection to send and receive voice data as against the traditional telephone technology that uses landlines to transmit analog signals. This has been mostly implemented through the connection of headsets to a personal computer or a special desk top IP phone. This study implemented a model that uses the normal android phones to support a phone system running on frequencyof2.4GigHz in a wireless local area network. The model was tested in Edo State Polytechnic campus and the results were satisfactory as touching the areas of voice quality and time latency under different conditions of LAN traffic.

INTRODUCTION

The traditional telecommunication infrastructure is facing a serious competition from internet protocol telephone system. The brisk development of the internet protocol in the past few years has consequently resulted in many aspects of internet technology such as voice and data communication systems. The traditional telecommunication infrastructure is no longer the only means for transmitting voice and data signals because the internet has provided a technology commonly referred to as voice over internet protocol (VOIP). Internet telephony has given users flexibility to make phone calls to others using the Internet Protocol, just as if they were using a normal telephone. This phone system that is supported by internet is providing a cost-effective way to make long distant calls which is very expensive with the normal telephone infrastructure.

The applicability of VOIP is assuming a wider dimension with the invention of android phones. The VOIP technology was before now limited to computers running popular desktop operating systems, but in the past few years special desktop phone set have been developed to support VOIP and the same support has been extended to android smart phones. VOIP can be used to send voice information digitally in discrete packets through any data network that uses Internet Protocol(IP), such as Internet, Intranets and local area network (Leu et al.,2014).

Edo State Polytechnic is a fast-developing tertiary institution. New buildings springing up with more laboratory. With this rate of development of the permanent site, offices are going to be more distant from each other, IP voice communication system will provide a cost effective and efficient means of communication for the campus community.

Review of Related Work

In the study done by the researchers in Design and Implementation of a VoIP Broadcasting Service over Embedded Systems in a Heterogeneous Network Environment, they used free software Linphone to practically establish the broadcasting architecture and evaluate the influence of audio buffer, jitter buffer in such a service. Then they proposed a scheme to broadcast voice packets to multiple embedded system-based receivers in heterogeneous networks' environments. A comprehensive experiment was conducted to verify the broadcasting service quality over heterogeneous networks, including LAN, 3G, and WiFi networks (Leu et al., 2014). The area of appeal of this emerging technology is the cost-effective way it provides voice communication around the coverage of the internet (Al Saedi & Salah, 2015).

There have been researches in this area focusing on different aspect of the VOIP technology and few of them are discussed in this section. In one of the studies reviewed the authors essentially examined the voice over internet protocol technology providing information on the architecture and underlying technology and the various types. They also presented relative description of the system and protocols of the VOIP technology. The use of compression algorithm in bandwidth management with the benefit and challenges of the VOIP technology also featured in this paper. The authors concluded that this technology can suitably become an alternative to the traditional telephone system (ALO & NWEKE 2013).

In a related research work (Al Saedi & Salah, 2015), LAN based IP telephone system was implemented in a computer lab using twelve personal computers (PC) connected to each other via TCP/IP LAN networks. NetMeeting SDK Active X control objects was used to support the communication process. IP address was assigned to all the computers individually to ensure that the message gets to the right place on the TCP/IP LAN network allowing the user to transmit information to a specific PC on the network excluding all the other PCs.

In another study the relevance of Voice over Internet protocol (VoIP) to the corporate environment was emphasized. The author also appraised implementation consideration as touching the most fundamental components of this technology and the quality-of-service assessment were discussed. They also gave a detailed survey of the entire VOIP technology and all the relevant equipment's required mostly at the user end of the entire network structure and architecture including some basic technical challenges associated with this technology. They concluded by looking at the implementation of this VOIP technology in a challenging satellite links (Jalendry& Verma, 2015).

This study Survey Paper on Voice over Internet Protocol (VOIP) appraised voice over Internet protocol (VoIP) functionality particularly on its support for users who wants to use the same data network to make voice calls. It was however remarked by the authors that the widespread deployment of this technology can elicit new security challenges, but its greatest potentials lie in the possibility of this technology to provide interactive communication services like video and voice conferencing(Shaw& Sharma, 2016).

The researchers in a different study showed how a royalty free audio codec Opus can be used as
VOIP in an interactive voice and multimedia application purpose. The applicability is described to be essentially in the areas of understanding mean opinion scale assessment when testing is been conducted to ascertain quality of service. Their observation showed that the change

associated with the various communication mediums and web real time communication (WEBRTC) in terms of protocol medium flow data types and rates of data service was highly insignificant. However, a high availability low-cost solution was provided for the analysis of big data from multiple service provider (Kaul & Jain, 2019).

Research Methodology

The development of a voice communication network on existing local area network (LAN) involved proper survey of the entire coverage area and thereafter a wireless network infrastructure was set up to serve as a tunnel for the voice communication. The configuration of internet protocol on all the communication devices that requires this initial setup. Caller ID's were configured on the communication server and on the CS simple software on the various android smart phones for the experiment. Voice communication is a sampling of an analog signal(verbal communication),its transmission is very sensitive to delays during transit. For voice to work correctly over the network infrastructure, the end-to-end transit time (cumulative time encoding the packet, leaving the sending client, traversing the network, and then being decoded at the receiving client) will be designed to match less than 150ms (milli-seconds). Issues encountered during transit result in imperfections in the reconstituted signal; also known as jitter. The jitter is basically the variation in delay that the system is experiencing. The study was divided into four phases. The phases are:

1. Coverage requirements and deployment planning
2. Network infrastructure and logical subnet design
3. Wireless "over-the-air" quality of service (QoS)
4. Voice client feature requirements

Phase One

Coverage requirements and deployment planning

In this phase, the project coverage area was provided which was critical to settings expectations for end users. The coverage of the network is determined to know the proper architecture of network infrastructure to be constructed for effective voice communication.

Phase Two

This phase involves deploying a wireless local area network (WLAN) that is voice-services-ready, it is important to anticipate the mobile nature of voice clients and to focus on the minimum expectation that calls will not get dropped as users roam across a building or campus. This means that the network will be deployed with continuous coverage in areas where voice services are planned, for coverage areas ranging from just a floor of a building to complete campus coverage, both indoors and outdoors. Areas such as main lobbies, employee entrance areas, parking lots, courtyards, cafeterias and supply, storage, and cage rooms will need WLAN coverage when voice clients are deployed on the campus. Roaming is integral to voice services on wireless networks. The wireless voice client will be designed to be able to maintain its association from one access point to another securely and with as little latency as possible. To accommodate roaming, IP addressing schemes is considered before deploying wireless voice clients.

Phase Three

Quality of Service(QoS)Considerations.

Wireless LAN(WLAN) traffic is nondeterministic; channel access was based on a binary back off algorithm defined by the IEEE802.11standard and is by nature variable, because it is based on the number of clients that access the network. The Virtual local area networks (VLANs) are taken to separate voice traffic from data traffic. This serves two purposes: security and isolation of higher-priority voice traffic so that it can be dealt with using maximum resources. To Separate voice from data requires a minimum of two VLANs which was constructed and assigned SSID on the WLAN for each VLAN. Using separate data and voice VLANs enables specific QoS settings on all traffic on the voice VLAN to give it a higher QoS profile. The configuration will enable voice traffic WLAN to use platinum QoS and assign low bandwidth. Separating traffic by VLAN and using the QoS profiles for VLAN traffic reduces the chance of data clients crowding the voice WLAN and causing unnecessary traffic over head and delays.

Phase Four

Voice client feature requirements.

This phase required designation of internet protocol phone at different user terminal. For call admission which involves voice conferencing and calls to user group on the network and testing to meet specifications.

Inaddition,theexpectationoftheteamasregardsuserterminalwasinternetprotocolphones,but during the implementation it was discovered that existing smart phones with certain specifications could be made to become internet phones using CSip simple software. CSip Simple is a Voiceover Internet Protocol(VoIP)application for Google Android operating system using the Session Initiation Protocol (SIP). It is open source and free software released under the GNU General Public License.

There are four main modules in our system. They are: -

1). Registration, 2). Authentication 2). GUI module 3). Audio call

The entire research work is mainly based on two modules i.e. server software and client software. Each user must have the client software. Administrator will have the server software. To run the client software server software must be in running mode to serve the request of each of the client. These programs can communicate using socket programming. Whenever we run the client software socket connection is established with the IP address of the server and the port number is opened for the communication.

Performance Evaluation

Session Initiation Protocol (SIP)

The exchange of signaling and information control is necessary in the entire call setup procedure from the beginning to the end between the calling and called parties involved. Mobility in a wireless local area network add little complication to this technology. The prior knowledge of the capabilities of the different devices in the network and the seamless flow of information between them is required before the efficient full flow of information can take place. This is handled by the SIP application layer control protocol working alongside the existing other protocols. A discovery of each other and establishment of a seamless connection of the destination and source "user agents" based on shared properties using SIP.

Quality of Service(QoS)Parameters

The traditional fixed infrastructure telephone services are at a disadvantage compared to the flexible data handling capability of the data networks. Different types of objective and subjective measures can be used to measure the QoS for VoIP, such as: The Mean Opinion Score(MOS),asshowninTable1and then, jitter and the end-to-end delay. The MOS is calculated using a non-linear mapped R factor as shown in Eq. (1), below:

$MOS = 1 + 0.035R + 7x10^{-6} + [R(R-60)(100-R)]$ equ(1)

Where:

R=100–Is-Ie–Id+A

Is: voice signal impairment effects, Ie: impairment losses suffered due to the network and codecs, Id: impairment delays particularly mouth to ear delay and A:Advantagefactor(attempts to account for caller expectations)

Table 1 Listening effort and quality scale

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	Distorted output

The 'jitter' "is the variation in arrival time of consecutive packets" evaluated over a period. Lett (i)be the time transmitted at the transmitter and$\hat{t}$(i),the time received at the receiver, the jitter is then defined as.

$Jitter = Max 1 \le i \le n\{[\hat{t}(n) - \hat{t}(n-1)] - [t(n) - t(n-1)]\}$...............equ(2)

The ‘Packet end-to-end delay’“ is measured by calculating the delay from the speaker to thereceiver compression and decompression delays” . Det, the total voice packet delay, is calculated thus
Det = Dn + De + Dd + Dc + Dde……………………………………equ(3)

where Dn, De, Dd, Dc and Dde represent the network, encoding, decoding, compression and decompression delay, respectively.
The ‘Packet Delay Variation’ (PDV)is an important factor to consider in network performance degradation assessment as it affects the overall perceptual voice quality. Higher packet delay variation leads to congestion of the packets causing more network overheads causing further degradation invoice quality. The PDV is the variance of the packet delay, which is given by:
$PDV = \{\sum_{i-1}^{n}([\hat{t}(n) - t(n)] - \mu)^2\}/n$…………………………………..equ(4)
Where: µ is the average delay of *n* selected packets.
$i-1$

A quality level is said to be unacceptable when the R rating is below the threshold of 50.Listening quality (LQ) and conversational quality (CQ) is used to interpret the R-factor-based MOS.MOS-LQ is used to capture the MOS score derived by considering coding distortions and packet losses, while on the other hand MOS- CQ is derived by considering delay and loudness impairments in addition to the distortion impairments (Hoßfeld et al., 2015).

The subjective evaluation is adopted in this study which captures the new paradigm of quality of experience (QoE) instead of the traditional quality of service (QoS). Over a long time, spanning decades service quality in communications networks has been assessed predominantly in terms of QoS (Quality of Service) parameters, like packet loss rate, delay, jitter, bandwidth as already mentioned above. However, this has started changing in the last few years with the adoption of a more user-centric concepts of service quality referred to as quality of experience, this has been mostly described as how a user subjectively rates an application or a service(Reichl et al., 2018). Therefore, our evaluation follows this latest trend as shown below in the end-user subjective description of acceptance of rejection of the implemented VOIP in wireless LAN. Two hundred candidates with equal representation from the sex and age demographics were selected for this subjective assessment adopting the listening effort and quality scale as shown below

Table 2 Summary of quality scale/scole from all respondents

S/N	Sex of Respondents	Age in years	Excellent/5	Good/4	Fair/3	Poor/2	Bad/1	Total
1	Males	<30	31	15	3	1	0	50
2	Males	>30	27	16	5	2	0	50
3	Female	<30	42	5	3	0	0	50
4	Female	>30	30	18	1	1	0	50
Total			**130**	**54**	**12**	**4**	**0**	**200**
Percentage quality scale score			**65%**	**27%**	**6%**	**2%**	**0%**	**100%**

The data from the subjective evaluation of the quality of experience by two hundred users from male and female two age demographics shows a close subjective assessment of the wireless LAN voice over network protocol as shown in table 7 above.

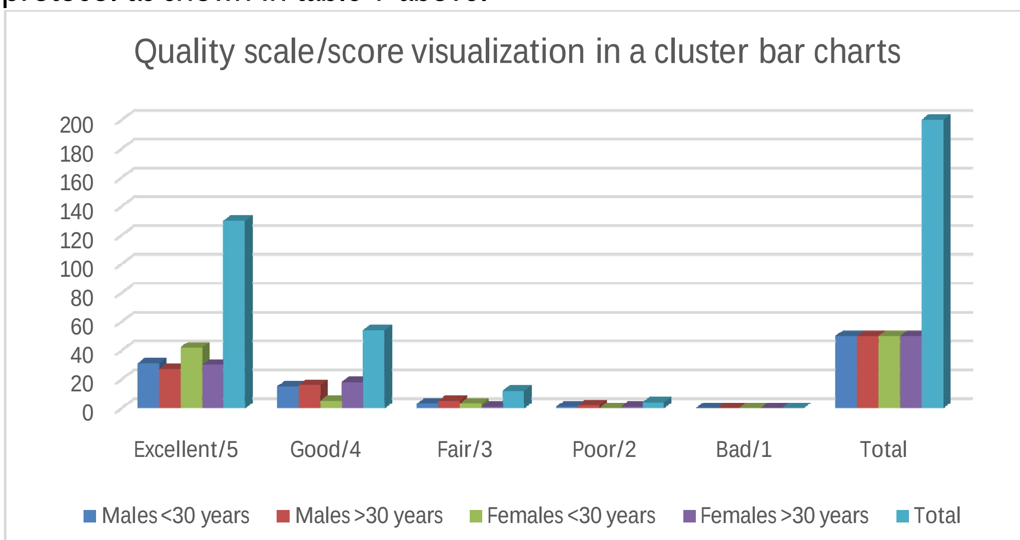

Figure 1 Quality scale/scole visualization in 3 D cluster bar chart

A visualization of the respondent's subjective evaluation report shows a consensus by 65% of respondents clearly adjudging the implemented system to be excellent, while 27% believes the system is good and having only 6% and 2% with a fair and poor experience, respectively.

Value added to knowledge.

Our system establishes communication between more than two participants at the same time which was not possible in previous similar work. In addition, our developments explore implementations for mobile platforms, teleconferencing as well as recording of calls which are novel in wireless local are a network powered voice communication. We have also added to knowledge in the area of line-of-sight consideration for wireless out door antennas, because in our work it was discovered that communication is possible without line of sight, this may be owing to the age of obstacles in the case of buildings or transmitting elements whose transmitting properties enabled them to send signals to other devices that were not directly visible to them, example of such obstacles are the roof of buildings.

RECOMMENDATION/CONCLUSION

This research work was about the development of VOIP over a LAN network in making a successful phone call within the coverage of the LAN without call charges. The system was tested using PCs, Allwox table sets and android mobile phones. The overall results of the system were found to be satisfactory for quality of the voice over the network. The SNR test shows that there is a very small difference between the SNR values during traffic and no traffic cases because the channel bandwidth is about 100 Mbps. We therefore recommend the full implementation of this system in our campus, for efficient and cost-effective communication.

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