

VoIP Limit Analysis on WLAN Networks

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Abstract – This work presents a study in progress about voice broadcasting on WLAN networks based on IP protocol (VoIP). This study proposes a latency analysis of these networks to broadcast VoIP considering implemented and non-implemented WDS (Wireless Distribution System) environments. In this context the study aims to identify mainly the problems that affect voice transmission on IP networks. VoIP sample calls are being analysed on both ambients proposed in this study. The data are captured through a packet analysing software on 100 samples in each ambient. With the usage of a language for graphical and statistical computing the partial results were presented through graphics.

Keywords— WLAN, WDS, VoIP

I. INTRODUCTION

The traffic generated by multimedia apps in real time grows on the internet exponentially day after day. It is very common to find mobile devices connected to the internet through wireless networks (WLAN), bluetooth or other wireless means. One of the multimedia apps used on this kind of network is voice over IP (VoIP).

Mobility provided by these devices is a current tendency. The need to access the internet anywhere combined with the possibility of keeping ourselves connected generates a new demand, the network availability and the services provided by it.

Making the access to WLAN network available and managing it within its reachability is one of the most frequent problems faced by the network administrators. The number of access points controlled may rise significantly depending on the network reachability according to the number of users to be integrated to the network. One of the technologies used to perform the WLAN wireless network distribution is WDS (Wireless Distribution System).

WDS allows the interconnection of access points without wires or any other infrastructure. This characteristic is very interesting when the aim is to expand to regions where it is impossible to reach with wire. When the function is implemented the access point begins to act as the client's signal repeater to the central access point connected to the main network. There the WLAN signal coverage rises or with the WLAN signal coverage, it rises according to the installation of more access points with WDS. Otherwise, the disadvantage is the reduced bandwidth due to the connected users sharing the network and a small latency is inserted into it.

Latency is caused when data packets take a greater time than expected to reach their final destination. In VoIP, latency causes disturbance like audio quality drop, voice delay, consequently the conversation cannot flow normally among the involved parts.

VoIP has characteristics of intolerance and delays, packet losses, not enough bandwidth, voice compression and even security. Packet loss can be considered one of the biggest problems of VoIP. It is associated directly to the call quality. With packet loss the user realizes clearly voice "cuts", that is, packet loss ends up undermining the quality of the call. Codecs used to compress voice signals have bandwidth requirements to work. If this bandwidth is not available for such application packet delivery delays may occur. With bandwidth shortage one of the main factors that harm directly is latency. A difficulty related to the security that may be mentioned is related to the DoS (Denial of Service) attacks, with the flood of connections attempts or unformed packets. As the most part of the residential networks does not have set up protection tools, this kind of attack may end up harming VoIP service leading even to its unavailability.

Many other worsening already known may be highlighted when VoIP use is related to other transmission means, mostly when the used mean is the wireless pattern.

This assignment has the objective to analyse the impact of VoIP implementation in WLAN networks, with the use of WDS through call analysis. To reach the research objective, a VoIP environment is going to be created in which initially many calls are going to be executed and each one of them is going to be processed with the use of packet analysers, in which it is going to be possible with the use of statistic tools to obtain the QOS's metrics involved in sceneries with WDS. What we try is to know the impacts caused by the use of WDS within VoIP service.

To evaluate such impacts we are going to do tests aiming to obtain QOS's metrics: delay, throughput, jitter and loss. From the obtained results, we hope to conclude that network metrics have different values in ambients with the use of WDS, which may compromise the quality of the service.

This assignment is organized in the following way. In section 2 the bibliographic study related to theoretical support of the research is going to be described. In section 3 the proposal of the study is described and in section 4 the used tools, the environment implementation of VoIP process used in the tests, methodology and data analysis are described. In

section 5 the preliminary results are exposed. In section 6 the future studies of the research are presented.

II. VOICE OVER IP IN WLAN NETWORKS

Local wireless networks (WLAN) are a viable and economic alternative to the conventional networks based in wire. The great number of mobile devices offered in the market having wireless interface grows every day, its pursuit is due mainly to electronic components price falls. Today it is possible to find different devices with wireless communication interface due to its setting facility, low-cost implementation and the mobility this interface provides for the device.

WLAN networks provide the same basic functionalities of cable networks (LAN), but combining mobility with speed access which results in the creation of new habits, relations and forms of work. A good example lies in the great usage of services that demand real time requisites, such as voice over IP applications and video in mobile devices.

The use of multimedia services (voice and video) may demand a reserve of resources and availability so it may have a good working and provide a good use experience to the user. Initially WLAN networks, based on 802.11, were not projected to this kind of traffic. For these multimedia services to work satisfactory, the associated use of packet prioritization techniques may be necessary.

The pattern WLAN 802.11b and 802.11g may support theoretically 200 and 900 VoIP sections respectively, but due to the inclusion of a great number of headers, this number is drastically reduced to just a few sections. The headers added to the payload consist on real time protocols (RTP), UDP, IP and headers from the MAC layer 802.11. According to [1], 802.11g pattern presents the best performance for VoIP traffic towards 802.11a and 802.11b, in both cases CBR (Constant Bit Rate) and VBR (Variable Bit Rate). A comparative study was conducted using 802.11a, 802.11g and 802.11b where the maximum number of sections and transmission rates on the Nominal and Multiplex-Multicast (M-M) regimes were illustrated, for both types of traffic CBR and VBR.

As an alternative to increase the number of sections supported in WLAN networks, the use of the M-M regime to send packets was proposed where. In this way the UDP, IP and RTP headers are compressed and rewritten in a mini-header and then combined in a single Multiplex-Multicast packet, where right after it sent through WLAN to the Multicast IP address.

Among the three patterns analysed in this article the 802.11g had the best performance compared to the others, with VBR traffic and M-M regime, the difference between the others appeared to be very significant in the number of VoIP sections supported and in the transmission rate reached. Using VBR traffic the tax flow is variable and there is the economy on packet sending with silence periods, which guarantees a better audio quality transmitted.

Another related problem to VoIP transmissions using WLAN networks can be described as the interference of many other networks that are very close, inducing the interference among them. When there are many other WLAN networks in

a small geographic area the number of supported sections is even more reduced, reaching values between 1,63 and 10,34 sections only using the GSM 6.10 codec.

According to [3] one of the main factors for the decrease in VoIP section numbers in areas with many WLAN networks is the mutual interference of CSMA (Carrier Sense Multiple Access). Many transmission sources using the same frequency or very close frequencies end up enduring each other interferences. This way the percentage of lost packets may increase, what is not interesting to any loss intolerant application, including VoIP.

To lessen the number of sections in Chan (2008) a Call Admission Control was proposed based on a graphic model of conflicts, where it was possible to rise in 52% the number of VoIP sections using the pattern 802.11b and 802.11g, which represented the number of 2,48 and 14,14 sections. The use of three orthogonal channels on WLAN was also proposed to reduce the lateral interferences, which raised the sections supported capacity on 802.11b and 802.11g patterns to 7,39 and 40,91. With CoTDMA (coarse-grained time-division multiple-access) use in partnership with CSMA it was possible to obtain a 35% improvement on the used patterns, which resulted in 10 sections to 802.11b and 58 sections to 802.11g.

The articles studied suggested alternatives to solve or improve the performance of VoIP usage in WLAN networks. In this assignment, as described previously the usage of WDS will be analysed as a form of WLAN network expansion, thus seeking to prove its performance in relation to VoIP and the impact caused.

III. PROPOSAL

This assignment proposes to realize bi-directional to VoIP clients with the objective of analysing the packets received by one of the clients, this way we can obtain information that are going to be used to calculate the metrics.

A. Testbed

Currently only some data have been processed and others are still in processing stage. Picture 1 illustrates the ambient defined with the school counsellor for sample collection

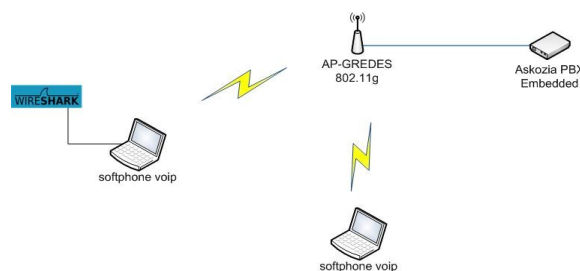


Figure 1. Ambient without WDS.

In the first testbed it was established that 100 calls should be made between both VoIP clients then capture the respective packets of each layer, 40 calls must be made in the morning, 40 in the afternoon and the remaining 20 in the evening. Each captured call must be saved in a unique pcap file, being

processed to the necessary information filtering for the statistic calculations. The calculations that must be done are based on the QOS's metrics: delay, loss, jitter and outflow.

The second testbed differs from the first, according to picture 2, because in this one it is going to be mounted a WLAN ambient using WDS. Basically five access points (AP) will be inserted with WDS function implemented to the main access point. The amount of sample calls is going to be the same as the previous testbed, the capture process will be made in different points of the network and the analysis processing will be the same.

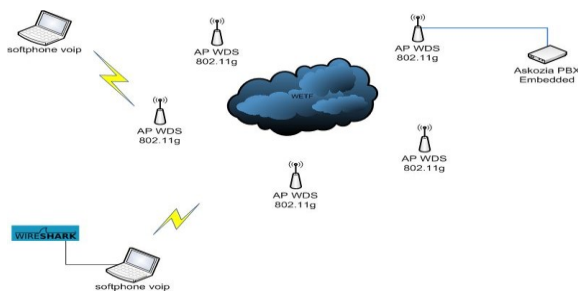


Figure 2. Ambient with WDS.

With the samples collected, processed and with respective graphics generated we are going to have the capacity to analyse the impacts caused by the use of WDS in WLAN networks in VoIP transmissions. The QOS's metrics will be analysed, compared and discussed, so that we get to a conclusion about the benefits of the use or not of WDS as a way of a WLAN network expansion, and mainly to point out the positive and negative points the use of this technology.

IV. METHODOLOGY

The bibliographic study performed on the initial phase of this assignment had the objective to underlie the arguments to be used on the testbed and aggregate knowledge on the technologies involved in the research. Studies in many areas related to the use of VoIP in WLAN networks were done, including security problems, performance, service quality, infrastructure, among other areas.

It is all about a quantitative research. Data are collected through the use of tool called Wireshark [7] installed in one of the computers. This tool has the objective to capture and save all packets transmitted and received by the wireless interface in a text file. The program basically captures entry and exit traffic and shows the information in an organized way. With the use of filters it was possible to customize the data that are of interest for the research, among them: packet size, average time, protocol, origin and destination, sequence of the packet.

In the proposed ambient the Softphone SFLPhone [6] was also configured in two computers of the laboratory of the Computer Network Research Group. Softphone made it possible to make calls directly among the computers. The codec used was ITU-T G.722, and the signalling protocol SIP (Session Initiation Protocol). Each made call had the average length of one minute among computers.

The WLAN network was mounted using an Access Point 802.11g without cryptography and connected directly to a PBX built-in server installed in a PCEngines's routerboard [5]. The Askozia PBX distribution [2] was used as an Asterisk telephone system.

The sample from the first testbed was composed by 100 calls made among the computers and captured on wireshark. Each made call was saved in a distinct file. Right after calls were made each file was converted to the CSV format (comma-separated values) where a filter was applied splitting the information of interest for the research. Based on this information the treatment and the data analysis were made with the aid of statistical program R.

With the filtered data of each call imported in R, it was possible to apply an exploratory analysis of the data through average descriptive statistics, median, standard deviation in each one of the imported files, as well as in general to obtain metrics between all made calls. The obtained data through the R tool are going to be illustrated in section 5, preliminary results.

V. PRELIMINARY RESULTS

The practical proposal of this assignment is to obtain bidirectional measures to VoIP clients with the objective of analysing the packets received by one of the clients, this way we can obtain the information that will be used to calculate the metrics. The delay was the metric processed until this moment in the assignment. The delay was obtained subtracting the packet time by its predecessor. In wireshark it is possible to obtain this value through delta_time column. In the R software each one of the correspondent files of a call was processed and the average delay of each one calculated.

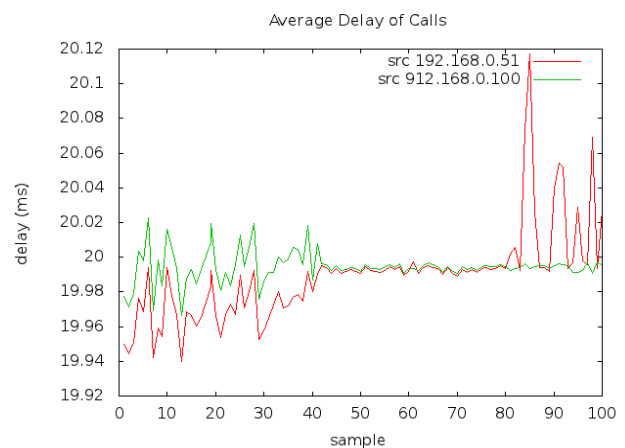


Figure 3. Average delay of calls.

The average delay of each call varied a little, it ended up with very close values, it is visually to observe that the delay variation among calls is quite small, considering the graphics scale.

Executing the function summary in the R software it was possible to obtain a summary from the average values:

TABLE 1. METRIC DELAY

Direction			
	V_MIN	V_MAX	V_MEAN
ip.src 192.168.0.100	19.97 ms	20.02 ms	19.99 ms
ip.src 192.168.0.51	19.94 ms	20.12 ms	19.99 ms

It was verified that the average delay metrics found is way below the maximum values recommended by ITU-T G.114 [4] that is 150ms. This value was expected due to background traffic being null, because in testbed moments no other app that can generate contestant traffic was used. With this we can obtain the initial values of an ambient without any traffic. These values will serve as a base to the values found in ambients with WDS use.

VI. CONCLUSION AND FUTURE ASSIGNMENTS

Real time apps as VoIP are sensible to delay variations. According to ITU-T recommendation the 150ms delay is considered the limit to VoIP apps. The main objective of the assignment is to know the QOS's metrics in WLAN ambients and evaluate its impacts in the use of a real time application. For this evaluation to happen it is necessary to create an ambient, where we collect samples and process the data in an attempt to know the impacts provoked by WDS in these networks.

The use of R software for calculus and graphics showed up as a very agile and practical tool for such kind of study. Its application may be seen in many areas, especially in the analysis of great volumes of information, because with meager resources it was possible to obtain, early enough, numbers that other softwares would take or consume much more computational resources.

With these partially found results we are going to continue the studies and begin to calculate the other metrics. We expect to know the other metrics and compare the impacts of VoIP. We also expect that our results can contribute to the people interested in taking decisions on WDS implemented sceneries.

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