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Publication: in Proc. 2nd IEEE Int’l Conference on Wireless and Mobile Computing, Networking and Communications (WiMob 2006)
Vol.: -
pp.: -
No.: -
Date: Montreal (Canada). June 19-21, 2006

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Performance of a Multi-Interface based Wireless Mesh Backbone to support VoIP Service Delivery

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Abstract—In this paper we evaluate the performance of a wireless IEEE 802.11-based node multi-hop mesh backbone for the provision of VoIP services, in the cases when nodes are provided with just a single NIC and when they are equipped with more than one NIC and different codecs are used. In particular, we show that the hard delay and packet loss constraints demanded by real time services constrain the size of the wireless backhaul to a few nodes when single-NIC nodes are used. Furthermore, this constraint depends on the link rate. This limitation is due to overlapping Carrier Sensing ranges that generate additional delay due to contention, which results in a strong degradation of the overall quality of the conversation. For the scenarios tested, the utilization of multi-NIC nodes overcomes these problems by allowing keeping the delay and losses within acceptable values while permitting larger wireless backhauls in terms of number of nodes. In this way, the MAC 802.11 policy keeps on playing a prevalent role in the very actual trend to build ‘all-wireless’ architectures in spite of the limitations experienced in the single-NIC case.

Index Terms— Wireless Mesh Networks, Multi-Interface, VoIP, 802.11, Wireless Backbone, Multi-hop, Coverage, Capacity

I. INTRODUCTION

THE last decade has witnessed an explosive growth of the interest in wireless technologies. This has led to a deep development and boosting of a wide range of research areas related to wireless networks. The perspective of connecting wirelessly to the Internet was tantalizing for the users and business attractive for service providers, so that the so-called Hot Spots began to be part of the everyday life even in the very early times of wireless networking research.

On the other side, the hourglass model that characterizes the TCP/IP stack gives IP a preeminent role in the current networking scenario. The flexibility and wide deployment of this protocol and the current demand for ubiquity convert this protocol in the most suitable one for the integration and convergence of well-established and emergent networks.

Within this scenario and boosted by the growing market interest, the primitive standard IEEE 802.11 has been progressively adapted to the exigencies the users demand for: boosted by the increasing demand for a better management of the quality of service (QoS) lots of efforts have been made leading to the formulation of the version 802.11e; successively, the 802.11i tried to offer a solution to the problem of the security. Responding to the current research interest on wireless mesh networking, recently it has been constituted the task group TGs within the IEEE 802.11 standardization body. This group focuses on elaborating over the concept of Wireless Distribution System (WDS) to form self-configurable mesh-like multi-hop networks.

The possibility to deploy an ‘all-wireless’ architecture\cite{1} able to support data delivery to mobile users as well as a reliable forwarded traffic begins to be taken into account. These new wireless architectures have been experiencing a great success as part of the deployment of indoor enterprise networks and in the deployment of wireless communities\cite{2}\cite{3}. They are intended to substitute parts of traditional wired core networks and, eventually, to replace them. The origin of this trend lies in obtaining an ‘all wireless’ Distribution System (as defined in the IEEE 802.11 architecture) that aims at extending wirelessly the area covered by a WLAN deployment.

The current necessary step is enhancing the actual conception of wireless access with network-oriented architecture solutions strongly oriented to reliability, in order to counteract the traditional performance degradation problems related to the transit of a signal through a wireless medium: that is, providing the nodes with enhanced features at PHY layer – beamforming techniques or directional antennas in order to avoid as much as possible interference and packet losses by taking awareness of the conditions of the channels – as well as formulating new and enhanced topology-aware MAC policies to support QoS \cite{1}.

Wireless Mesh Networks (WMN) are a solution\cite{4} to merge all these constraints together and to provide the wireless segment with the demanded reliability. In Fig. 1, the reference architecture of a WMN is considered. In this architecture, the wireless routers (WR) forming the wireless backhaul play a
double functionality. On one side, they are in charge of routing the traffic between any two end-points of the network. On the other side, they act as traditional Base Stations (BS) providing access to mobile nodes. One of the challenges of WMN is the possibility to provide each node with multiple interfaces [1]. This appears to be an essential step towards the robustness of wireless backhauls, since it allows for a reliable radio meshing – studies about channel allocation have been carried out in [5] – and also promises to open new challenges as far as the handoff management is concerned.

Fig. 1 A Wireless Mesh Network reference architecture. Wireless routers form a wireless backhaul and act as traditional Base Stations providing access to Mobile Nodes.

In this paper, the performance of such a backhaul is evaluated. We compare the performance of a traditional single-interface based wireless multi-hop network with that of a multiple-interface based one when carrying voice over IP (VoIP) traffic. Once deployed, such a scenario is expected to be a real competitor to traditional cellular telephony.

We will point out that the overall degradation of the quality of the voice is due to delays introduced by the overlapping of Carrier Sense (CS) ranges.

The main contribution of this work is the assessment of the benefits coming from the deployment of multi-NIC-based backbone in terms of delay and packet losses; to the best of our knowledge this is the first study directly addressing the issue of Multi-NIC-based wireless networks for real-time service provisioning.

The rest of the paper is organized as follows: in Section II an overall description of the methodology used to evaluate the resulting QoS of a VoIP communication – the EModel – is provided, together with the description of some voice codecs actually in use. Subsequently, in Section III the whole simulation system is discussed, detailing the topology and the simulation set-up and stating the objectives of our analysis; in Section IV a detailed analysis of the results is provided and, finally, in Section V conclusions are drawn.

II. BACKGROUND

A. E-model

The Emodel, defined in ITU-T G.107 [6], is a computational model that predicts the voice quality of a phone call using transmission parameters. It gives an overall rating for the quality of a call, on a scale from 0 to 100, called the R-factor.

The Emodel combines different impairments based on the principle that the perceived effect of impairments is additive:

\[ R = Ro - Is - Id - Ie + A \]  

Ro is the signal to noise ratio, based on send and receive room noise levels, the circuit noise and the noise floor. Is includes impairments that happen simultaneously with the voice signal, such as sidetone and PCM quantizing distortion. Id comprises delay impairments, including impairments caused by talker and listener echo and by a loss of interactivity. Ie includes distortion of the speech signal due to encoding and packet loss. Finally, A is the advantage factor and represents the degradation in quality accepted by the user in return for the ease of access (e.g., when using cellular or satellite phone).

Fig. 2 The R-factor scale, ranging from 0 to 100.

Fig. 2 depicts that the R-factor equals to 70 can be considered as the minimum value such that the quality of the VoIP service can be considered acceptable (i.e. equivalent to that of traditional PSTN).

B. Codecs for VoIP

Codecs are used to convert an analogical voice signal into a digital one. There are many encoding schemes that have been developed and standardized: the simplest one is the sample-based G.711, which uses Pulse Code Modulation (PCM) and produces a rate of 64 kbps. It is a common reference for speech compression quality. Each sample of the original signal is encoded in 8 bits or an octet. It uses no compression and it is the codec used by the PSTN network and ISDN lines. The signal can be sampled in two different ways: -Law is used in the US and Japan and A-Law in Europe.

CELP-based encoders provide rate reduction at the expense of lower quality and additional complexity and encoding delay. One of these codecs is the G.729 one, which uses a CS-ACELP speech compression algorithm encoding 10 milliseconds of speech at a rate of 8 kbps. G.729 Annex A is a reduced complexity version of the G.729 coder.
In addition, there is also an extensive set of speech coding standards for digital cellular applications. In this way, the GSM standard supports four different but similar compression technologies to compress speech. These include full-rate (FR), enhanced full-rate (EFR), adaptive multi-rate (AMR), and half-rate (HR). One of them, the GSM-EFR, is an ACELP compression algorithm which encodes 20 milliseconds of speech at a rate of 12.2 kbps.

III. SIMULATION FRAMEWORK AND SCOPES

A. Topology and simulation set-up

The system we have considered is composed by a succession of 802.11-based wireless nodes disposed along a serial topology, as depicted in Fig. 3. This configuration is close to the reality of deployment of wireless backhauls in urban environments. Existing urban infrastructures are good candidates to be exploited by installing wireless routers (WR) on the top of, for example, the light poles [4].

Furthermore, each node is configured in such a way that can correctly receive only the messages coming from its own adjacent nodes, that is, if we define $A^\text{RX}_i$ the area in which the $i$-th node is able to correctly decoding an incoming packet and if $N$ represents the total number of the nodes, we achieve the relation (2)

$$A^\text{RX}_m \cap A^\text{RX}_n = \emptyset \text{ if } |m - n| \geq 2, \forall m, n \in \mathbb{N}: m, n \in \{1, \ldots, N\}$$

In this case, we have supposed $A^\text{RX}_1 = 50$ meters and the distance between two adjacent node has been set up to 40 meters.

The Ad hoc On-Demand Distance Vector Routing (AODV) is the protocol by means of which the nodes obtain routing information.

The first node (node 1) of the chain hosts a VoIP generator that transmits packets to the last node of the topology (node $N$) through the intermediate relays.

We have taken into account two different cases. In the first one, every node is equipped with a single Network Interface Card (NIC) with the same characteristics of those in other nodes, while in the second one, every node is equipped with two different NICs, each having its own transmission and reception pattern.

The system has been studied using the Network Simulator 2 (ns2) [9]. In particular, the wireless node model provided by the Monarch Group Project [10] has been modified in order to make the wireless node able to support more than one NIC.

B. Objectives of the study

The scope of the simulation study is to investigate the possible benefits coming from the utilization of more than one radio interface in the deployment of an ‘all-wireless’ backbone network. Different scenarios will be in turn considered:

1) Single-NIC node, ideal case: in this case a single VoIP connection streaming from node 1 to node $N$ is considered. All the nodes of the system are equipped with a NIC of the same characteristics. It is assumed that no packet losses are introduced in the system by the presence of the wireless channel.

2) Single-NIC node, real case: every node works with the same type of NIC. Packet losses are modeled taking into account the effects of the channel. In particular, given $N$ the number of nodes and given $\text{dim}_p$ the size of the packets flowing into the system (in bits), the overall packet error is assumed to be – see the APPENDIX for an explanation of (3).

$$\text{PER} = 1 - \left[1 - \left(1 - 10^{-5} \text{dim}_p \right)^{N-1} \right]$$

In both cases (1 and 2), the overall quality of the VoIP transmission is shown with respect to the Link Rate (LR). In particular, three values are considered, namely 2 Mbps, 11 Mbps, and 54 Mbps. Furthermore, the variations of the performance with the RTS/CTS mechanism enabled or disabled are also analyzed.

3) Single-NIC node, codec case: a comparison between the performance of a VoIP transmission when using different codecs is performed.

4) Double-NIC node, real case: every node is equipped with two NICs with the following conditions:
   - each node is able to receive on one NIC and simultaneously transmit on the other one.
   - both NICs in a node work in orthogonal channels. As a result, adjacent nodes do not interfere with each other.

The results described below are presented in terms of the quality of the voice at the receiver (evaluated by means of the R-factor) plotted versus the number of the nodes composing the topology. The R-factor is computed considering the joint effect of packet losses and average delay introduced when the number of nodes in the mesh backhaul increases.
IV. SIMULATION RESULTS

A. R-factor vs Network Size (the case for single-NIC wireless routers)

Fig. 4 depicts the impact on the R-factor of increasing the number of nodes in the network topology (N) when the codec G711 is used. It can be noted that the acceptable (i.e. R>70) number of nodes increases as the link rate (LR) also increases. This can be explained noticing that, as the LR increases, the system incurs in lower transmission times, thus reducing the effects due to the average point-to-point delay on the degradation of R. For example, it can be seen that moving from 2 Mbps to 11 Mbps allows a larger number of nodes to compose a more extended backbone (up to 7 nodes more).

It is worth noticing also that the R-factor decreases linearly up until a certain value when it drops dramatically. To explain this effect we plot in Fig. 5 the average end-to-end delay versus the number of nodes in the network. One can see that the delay increases linearly until it suddenly grows very fast heavily affecting the quality of the received VoIP signal (the R-factor suddenly drops). Again, the total number of nodes for which linearity is kept constant depends on the Link Rate (LR) configured in the NICs.

Although the delivery of a single traffic flow is not the case in which enabling RTS/CTS mechanism lead to advantages it is worth highlighting the degradation it introduces in terms of extension of the serial topology. Fig. 6 shows that if LR = 2 Mbps, enabling the RTS/CTS mechanism in all the NICs makes the maximum number of relays supporting quality VoIP calls to drop from 13 to 9 (we are losing 200 meters of network extension). The corresponding increase of the average delay is shown in Fig. 7.
When the channel is not ideal and introduces transmission errors, the quality of the voice initially decreases linearly before rapidly dropping to 0. Fig. 8 shows that this packet loss phenomenon only slightly affects the overall performance: in fact, before getting down the R-factor keeps on staying in a range (ranging from 92 to 85) where most users claim to be satisfied.

Moreover, neither it results in a heavy dropping of the number of nodes nor it amplifies the degradation due to the enabling of RTS/CTS. Fig. 9 shows that introducing the RTS/CTS mechanism keeps on providing the same drop for the R-factor already observed in Fig. 6.

The rest of this subsection discusses the causes of a R-factor breakdown in figures 6 and 8.

-Causes for the R-factor breakdown

More than one study in the literature have shed some light on the weakness of the IEEE 802.11 MAC in supporting transmissions over multi-hop wireless networks. In [11] the authors highlight how the capacity of a serial topology decreases as the number of nodes gets higher and higher, while in [12] the TCP instability over multi-hop networks is addressed.

In the above figures, a huge and sudden degradation of the R-factor is observed and, as a consequence, an unacceptable quality of the voice is perceived by the final user. Basically, the R-factor as defined in the E-model, depends on the codec being used, the average end-to-end delay, and the percentage of packet losses. The R-factor is very sensitive to variations of these parameters, specially to packet loss variations.

Actually, there are many causes behind the evaluation of a packet loss; basically, losses can be experienced as a consequence of channel errors, due to collision events (that can be somehow recovered by means of retransmission at the MAC layer), and finally, due to packet droppings at the transmission buffer of a node –as it is limited in size–. These three phenomena are taken into account by means of three different packet error rates, respectively $p_{ch}$, $p_{coll}$, and finally, $p_{DROP,IFQ}$ such that for a given PER the overall packet error rate can be expressed as

$$PER = PER(p_{ch}, p_{coll}, p_{DROP,IFQ})$$  \hspace{1cm} (4)

In Fig. 10 the component $p_{ch}$ is represented as a function of the number of nodes. Together, in the same picture, a joint representation of $p_{coll}$ and $p_{DROP,IFQ}$ is also represented as a function of the number of the nodes. The picture shows that while $p_{ch}$ grows linearly, regardless of the number of nodes there exists a moment in which the jointly effect of collisions and packet drops in TX buffers become prevalent and heavily leads the behavior of the resulting PER'.

We have observed that this last phenomenon is mainly due to the fact that although each node can only transmit to its adjacent ones, the Carrier Sense Range (CSR) of each of the nodes is always larger. We define the CSR as the area where a given mobile node can listen to the ongoing transmissions between couples of nodes consequently perceiving the medium as busy, thus deferring its own transmissions for avoiding collisions. As a result, nodes end up competing with relays that are beyond their transmission range but within their carrier sense range and unexpected transmission delays and losses appear.

In addition to this phenomena, we have observed that the number of retransmissions (i.e. the retry limit) in case of a collision even worsens the impact of the breakdown of the R-factor (as it increases the number of losses and the end-to-end delay).

Fig. 11 plots together the R-factor and the losses of the system and shows how the main contributor to the R-factor breakdown are the losses caused by collisions and transmission buffer overflows while the contribution of channel errors is not significant.
Fig. 10 Packet loss nature comparison. It worth noting that although the channel error trajectory appears to be linear, it is actually given by the power of the number of hops (see the Appendix).

Fig. 11 Effects of Packet loss on the overall performance

B. R-factor vs Network Size (varying the Codec)

In Fig. 12, the influence of the usage of different codecs on the R-factor has been considered: it can be noted that the G711 is the one able to offer the best performance.

On the other side, for not very large topologies the usage of GSM seems to be a better solutions in comparison with the one provided by the G729, although it shows to be more sensitive to packet losses, since the slope of the corresponding curve is higher than the G729’s one.

C. R-factor vs Network size (the case of MultiNIC Wireless Routers)

This section shows the benefits coming from providing every node with the possibility of using two different wireless NICs. Hitherto, we have seen that the usage of a single-NIC based nodes leads to a breakdown in the overall performance once a critical size is reached. The conditions for the breakdown depend on various factors, such as the number of nodes, link rate (physical modulation) used, etc. Fig. 13 shows how this problem can be overcome when using multi-NIC nodes and decoupling the reception of the transmission (i.e. transmission and reception of each of the two NICs in a single node is done through orthogonal channels).

Fig. 12 R-factor vs N for different codecs

Fig. 13 MultiNIC enabling benefits on VoIP performance.

In this manner, the multiple collision domains of the single-interface case are turned here into a set of orthogonal receiving/transmitting domains. The behavior of the R-factor keeps on being linear for a growing number of nodes in the network while it would collapse in the case of a single-NIC
based network, for the tested sizes. In Fig. 14 we report the same phenomenon, but this time analyzed from the average end-to-end delay point of view. In this case, we find that in the multi-NIC case, the average end-to-end delay is proportional to the number of hops for the tested sizes. 

Let us define \( p_e(i), i=1,2,\ldots,N \) (see Fig. 15) the probability for a packet to be considered as error-affected by the generic node \( i \), and \( p_{ehp} \) the resulting packet error probability characterizing a single hop; the probability to have no errors during the transmission through a generic hop is given by \( (1- p_{ehp}) \), while the probability to have no errors at node \( k \) is given by the probability to experience no errors during \( k-1 \) hops.

![Fig. 15 The multi-hop topology suggests the idea to apply the Bernoulli’s experiment to evaluate the resulting Packet Error Rate as a function of the number of hops.](image)

That is a Bernoulli experiment and the probability \( p_e(k) \) to have no errors at node \( k \) is given by

\[
p_e(k) = (1- p_{ehp})^{k-1}
\]  

Consequently, every node is characterized by a Packet Error Rate \( p_e \) as a function of \( k \) and given by:

\[
p_e(k) = 1 - p_{ne}(k) = 1 - (1 - p_{ehp})^{k-1}
\]  

Finally, the PER at the end of the considered chain is given by

\[
PER = 1 - p_{ne}(N) = 1 - (1 - p_{ehp})^{N-1}
\]  

The last step is to give an estimation of \( p_{ehp} \). Assuming that a single error on a single bit is able to affect the correct reception of the whole packet, we can consider the equivalence

\[
p_{ehp} = Pr(\text{at least 1 wrong bit}) = 1 - Pr(\text{No bits are wrong}),
\]

and since \( 10^{-5} \) is assumed to be a target Bit Error Rate (BER) for 802.11 technology, the probability to have no errors for a packet of length \( dim \) is once again a Bernoulli trial and given by

\[
(1-10^{-5})^{dim}.
\]

Merging together this last relation with the Equation (a3), we obtain

\[
PER = 1 - \left[1 -\left(1 - (10^{-5})^{dim} \right) \right]^{N-1}
\]  

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