UNIVERSITY
OF TRENTO

DEPARTMENT OF INFORMATION AND COMMUNICATION TECHNOLOGY

38050 Povo – Trento (Italy), Via Sommarive 14
http://www.dit.unitn.it

COST-EFFECTIVE VOIP SERVICES FOR REDUCING DIGITAL DIVIDE IN DEVELOPING COUNTRIES: CASE OF STUDY AND PRACTICAL IMPLEMENTATION

C. Sacchi, M. Piazza, , and FGB De Natale

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Cost-Effective VoIP Services for Reducing Digital Divide in Developing Countries: Case of Study and Practical Implementation

Claudio Sacchi, Matteo Piazza, and Francesco G.B. De Natale

Abstract

Digital divide is one of the most relevant emergencies of developing countries. The situation is particularly difficult in central Africa, where wide zones are totally disconnected by any kind of communication services. In these areas (sometimes not very far from big towns) many people are living and critical infrastructures, like e.g. hospitals, cannot communicate with the external world. In this work, we are going to propose a feasible and low-cost solution for installing VoIP-based telephone services in peripheral areas not reached by fixed and cellular networking infrastructures. The proposed solution is based on a Wi-Fi radio bridge connecting a remote disconnected site with a gateway placed in a town. The gateway consists only of a PC connected to the PSTN by a suitable interfacing card. The gateway is equipped with the ASTERISK server, a software tool able at routing VoIP calls to a PSTN, as it would happen if a regular push-button telephone was employed. The system was developed and tested in open field and provided very good results in terms of efficiency and reliability. Such a positive testing phase encouraged developers to propose such kind of solution also for practical installation in interested countries. In particular, low hardware costs, easy of use and totally open-source software availability are regarded by potentially interested users as real strength points with respect to other available commercial solutions.

I. INTRODUCTION

IN a global interconnected World, words like “total coverage”, “ubiquitous services”, “broadband access”, “nomadic wireless access”, etc. represent the common trend towards a deeper and deeper penetration of digital communications in the everyday reality. But, such a perspective becomes purely virtual, when we are considering developing countries of Africa, part of Asia, and Latin America. In these countries, millions of people cannot benefit by digital services. Truly, they cannot benefit by any kind of telecom service at all, because neither cable, nor antenna, nor fibre is installed to bring them some Hz of bandwidth. This is the problem of so-called *digital divide*. As stated in [1], the digital divide exists among countries (the global divide) and within countries (the social divide). There is an overall trend of a growing divide on both fronts. Statistics about the penetration of telecommunication services reported in [2] evidenced a number of telephone lines per inhabitant in Central Africa equal to 0.016. The corresponding numerical datum is 0.661 for USA and 0.287 for Japan. Behind this numbers, there are many local situations that seem incredible for people living in developed countries. Our research group working at University of Trento cooperates with an Italian no-profit organization: “Ingegneria Senza Frontiere” (ISF), that manages projects targeted to the provision basic infrastructures for people living in developing countries. About two years ago, ISF was contacted by the personnel of a big Nigerian hospital located fifteen kilometres far from the town of Owerri in the Imo State. A guy working in this health infrastructure (hosting hundreds of in-patients and provided with modern medical equipments) told to ISF that they couldn’t dial by phone with anyone, because they have neither PSTN connection nor cellular coverage. PSTN and GSM network are fully available at Owerri, but not close to the hospital. The local telecom company doesn’t forecast to install telephone lines and/or base stations outside Owerri metropolitan area. So, the hospital should take in charge all the big expenses for the installation of a dedicated telephone line. At the end of the story, the Department of Information and Communication Technology of the University of Trento accepted to investigate, in cooperation with ISF, the feasibility of a low-cost solution in order to solve the problem mentioned by the Nigerian hospital. In this paper, we are going to report substance and results of the R&D study carried out.

In particular, we studied, implemented and tested a cost-effective solution based on three basic technological points:

- Use of Voice over Internet Protocol (VoIP) [3] as application layer;
- Use of IEEE 802.11b standard [4] for point-to-point transmission of voice packets between a remote site and a building hosting a PC connected to a PSTN;

- Use of ASTERISK server open source software [5] in order to route the VoIP call on the PSTN by using a proper interfacing card.

Having a look to the state-of-the art about VoIP services and Wi-Fi, we can observe that the provision of VoIP over WLAN have been considered in [6]. A Wireless Distribution System (WDS) has been employed in [6] in order to connect the WLAN to the wired backbone. In [7] a broadband wireless access system for rural areas has been presented. The aim of this system is to provide broadband and VoIP services to people living in isolated and disconnected rural areas of Montana (USA). IEEE 802.11b standard link is employed in the local loop, whereas the bridging is performed by a point-to-point 10-Base-T wireless link. The traffic is then conveyed to Internet by a wired backbone placed in a metropolitan area. Voice performances in WLAN networks have been assessed in [8] by means of an experimental study. In particular, voice services have been tested in the presence of background data traffic, providing an evaluation of the capability of 802.11b MAC layer to support simultaneous voice and data. From some points of view, our setup is quite similar to the one proposed in [7]. But in our case, the system is only targeted to the VoIP service. Our claimed aim is to replace, with a moderate expense, a fixed telephone line by using a wireless low-cost communication infrastructure. The main innovation of the present work mainly consists in the application context (i.e.: digital divide reduction) and in the use of Wi-Fi to replace twisted pairs. The paper is structured as follows: Section II contains a global description of the system. Section III will be focused on the procedure for call routing from peripheral VoIP terminal to the PSTN. The experimental setup will be shown in Section IV, together with some experimental results. Finally paper conclusions are drawn in Section V.

II. GLOBAL SYSTEM DESCRIPTION

In this section, we wish to provide readers with a global description of the proposed wireless-based VoIP system, together with an explanation of the main reason that motivated us to select some technical solutions instead of other ones present in literature and/or in commerce. The technical details of the system components employed for the actual implementation will be described in Section IV. In the meanwhile, a global overview of the system is drawn in fig.1.

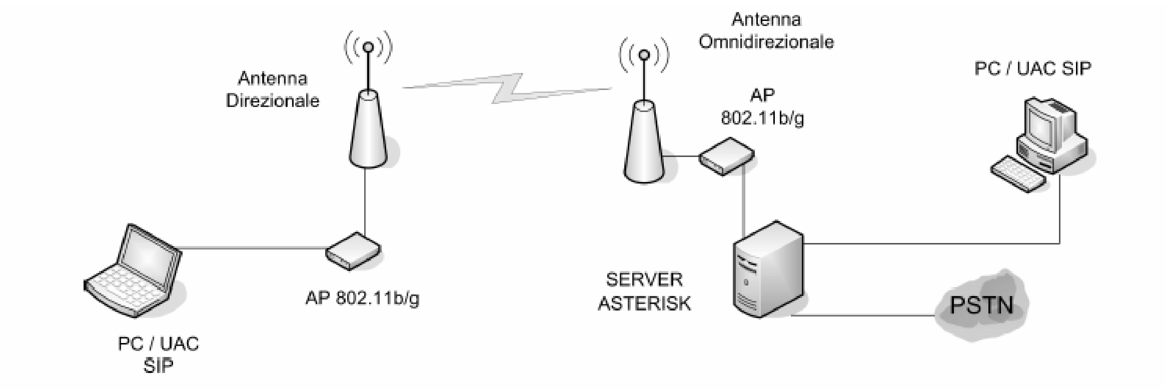

Fig. 1 Global scheme of the proposed VoIP system

The VoIP terminal of the remote disconnected site is substantially a PC or a laptop, where the open-source Session Initiation Protocol (SIP) [9] software has been previously installed. The choice of SIP protocol instead of the standard H.323 protocol was mainly motivated by the ease of use (H.323 is more complex to be managed) and the ease of interfacing with the ASTERISK server.

The VoIP terminal transmits voice packets via wireless using a WLAN IEEE802.11b/g connection. The choice of IEEE 802.11b/g standard working in the 2.4GHz ISM band is compliant with the aims of the proposed research work. In fact, the usage of such a bandwidth is allowed, without need of licences and permissions, in the largest part of African continent [10] (also in Nigeria, where the final system release should work). In this work, we considered and experimented two alternative configurations for the WLAN connection:

- A conventional IEEE 802.11b/g connection using a wireless PCMCIA card installed at the remote site that communicates with the AP located at the gateway site;
- A Wireless Distribution System (WDS) connection modality [4] that interconnects two access points: one located at the remote site and the other one at the gateway site. The employment of WDS instead of conventional IEEE 802.11b/g access modality has been regarded by us as an effective solution in order to enhance connectivity and reduce jitter in long-distance bridging.

In both cases, the wireless transmission to the gateway is performed by using a directional antenna. At the gateway site, we have an omni-directional antenna. The usage of an omni-directional antenna at the receiver side is very suitable for our application, because it can provide high gains with reduced dimensions and it is shaped for easy outdoor installations. Moreover, an omni-directional antenna allows us to cover a wider area and to provide a unique connection point for different remote transmitting sites. Finally, the adoption of a couple directional-omni-directional antennas allows us to easily pointing the two antennas without using specific testing instrumentation. Inside the premises of the gateway site, a PC is interconnected to the PSTN by means of the ASTERISK software. ASTERISK is an open-source software written in C language that can support the functionality of VoIP gateway. It can be employed in several applications [5]:

- Heterogeneous Voice over IP gateway (MGCP,SIP,IAX,H.323);
- Private Branch eXchange (PBX);
- Custom Interactive Voice Response (IVR) server;
- Softswitch;
- Conferencing server ;
- Translation;
- Calling card application;
- Predictive dialer;
- Call queuing remote agents;
- Remote offices for existing PBX.

ASTERISK practically works as a middleware layer placed between the PSTN infrastructures and the application layer. As far as VoIP services are concerned, the following protocols can be effectively supported by the ASTERISK platform [5]:

- Session Initiation Protocol (SIP);
- Inter-Asterisk eXchange (IAX) version 1 and 2;
- Media Gateway Control Protocol (MGCP);
- ITU H.323;
- Voice over Frame Relay (VOFR).

In Fig.2 the logical diagram of ASTERISK server architecture is depicted.

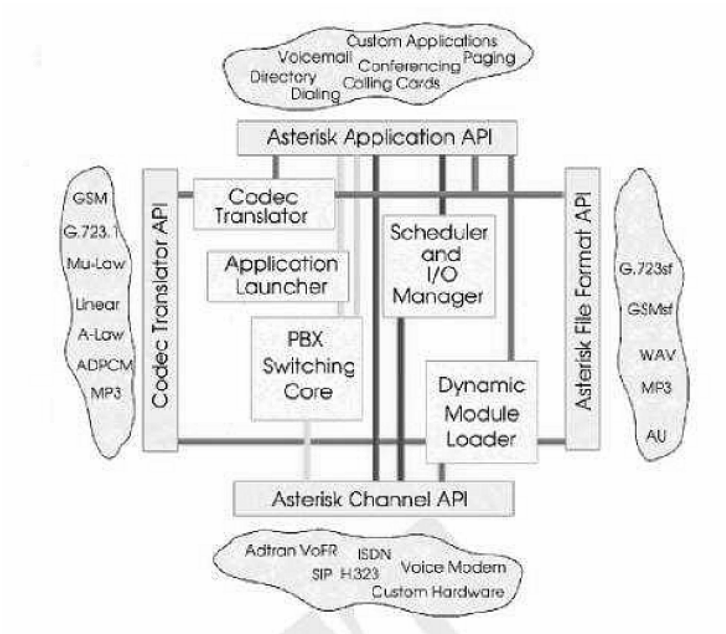

Fig.2. ASTERISK server architecture

When ASTERISK is switched on, the *Dynamic module loader* memorizes and initializes all the parameters needed for connection, like e.g.: the VoIP protocol adopted, the active telephone channel, dial plan details, employed codecs and links them with the appropriate internal Application Program Interfaces (APIs). Afterwards, the *Switching core* starts to accept the calls coming from the different interfaces according to the dial plan. The *Application launcher* is devoted to make physically ringing telephones, connecting vocal boxes, etc. Finally, the *Codec translator* allows connection to voice channels using different codecs. In the next section, we shall describe how such tools have been employed in the proposed system in order to send a phone call from source to destination. The hardware interface between the ASTERISK server and a PSTN is a Foreign eXchange Office (FXO) card. This hardware component physically manages the basic operations of call initialisation and call reception issued by the ASTERISK server.

III. VoIP-BASED DIAL PROCEDURE

The VoIP packets are sent from the remote site to the urban gateway using a UDP/RTP transport protocol [11] without retransmission of the wrong packets. Such a solution has been considered in order to avoid latencies that would be involved by TCP/IP when the system is waiting for a retransmitted packet. RTP unavoidably losses wrong packets and this fact obviously impacts in some way on the entire system performances, as shown in Section IV. The VoIP-based dial procedure can be summarized as follows (see Fig.2 where the signalling flow from the remote VoIP terminal to the

ASTERISK server and to PSTN is drawn):

i) The remote VoIP terminal sends the invitation message *INVITE* to the ASTERISK server. This message contains the call request, together with the address of the destination terminal. Users can exploit the simple addressing mechanism of SIP that is very similar to the e-mail addressing modality. For example, if a remote user wishes to send a call to a given destination number, he must write an address command in the following form:

sip:*<phone number>@<Gateway ID>*.com

ii) ASTERISK sends the "waiting" message *TRYING* to the requesting remote terminal. At the same time, it converts the SIP message *INVITE* into an "initial address message" (IAM) that is sent through the PSTN (the standard signalling protocol ISUP is employed by ASTERISK for communications with PSTN [5]);

iii) ASTERISK receives an Addressing Complete Message (ACM), confirming that the destination terminal has been contacted and the communication channel reserved for the call. Then ASTERISK immediately sends a *RINGING* message to the remote terminal that alerts it about the established connection;

iv) The destination terminal accepts the call (i.e.: the destination user "hangs up" the telephone receiver). When this happens, an "Answer Message" (*ANM*) is sent through the PSTN to ASTERISK. ASTERISK forwards an OK message to the remote VoIP terminal that answers to ASTERISK by sending an ACK message;

v) The conversation starts. Gateway and VoIP terminal exchanges audio packets. The VoIP terminal sends audio packets that are appropriately converted by the Gateway:

    a. In an analog voice message to be sent through the PSTN, if the destination terminal is a classical push-button telephone;

    b. In a suitable voice packet format, if another VoIP terminal works at the destination side.

vi) The conversation ends. The SIP message *BYE* is converted by ISUP into the "release" *REL* message sent to the destination terminal. The "release complete" *RLC* message points out from the PSTN side the "hanging down" of the telephone receiver. The last message is the SIP *OK* message sent by ASTERISK in order to inform the remote VoIP terminal that the dial is really closed.

```
SIP                         Asterisk                      PSTN
 1|---------INVITE---------->|                            |
 2|<------100--TRYING--------|                            |
  |                          |------------IAM---------->|3
  |                          |                            |
  |                          |<-----------ACM-----------|4
 5|<------180-RINGING--------|                            |
  |                          |                            |
  |                          |<-----------ANM-----------|6
 7|<------200--OK------------|                            |
 8|-----------ACK----------->|                            |
  |                          |                            |
  |<========AUDIO==========>|<==========AUDIO=========>|
  |                          |                            |
  |                          |                            |
  |                          |                            |
 9|-----------BYE----------->|                            |
  |                          |-----------REL----------->|10
  |                          |                            |
  |                          |<----------RLC------------|11
12|<------200--OK------------|                            |
```

Fig.3 VoIP-based dial procedure: a flowchart

IV. EXPERIMENTAL SETUP AND RESULTS

A. *Experimental setup*

A demonstrator of the proposed wireless-based VoIP system has been developed and tested in open field. Different test sites have been considered in the urban and sub-urban areas surrounding the city of Trento. Such a choice has been motivated by the impossibility of installing in short time the experimental setup really on-site (i.e.: in the premises of the Nigerian hospital close to Owerri). Nevertheless, people working in this health infrastructure and other people working at the Federal University of Owerri have been contacted during the development of the project and timely informed about the achieved results. In Tab.1, the geographic coordinates of the installation sites are reported. The installation site #1 hosts the ASTERISK server and the PSTN connection. The installation sites #2,..,#6 correspond to remote VoIP terminations placed at different distance far from the gateway.

TABLE 1.GEOGRAPHICALCONFIGURATION OF THE TEST SITES (SOURCE: CARTOGRAPHIC SYSTEM OF TRENTO PROVINCE)

Site #	Coordinates	Altitude	Distance from gateway
1	11°07’06,469’’ 46°13’00,044’’	210 metres.	-
2	11°07’06,923’’ 46°13’03,873’’	215 metres.	100 metres
3	11°06’12,607’’ 46°11’34,857’’	210 metres	2779 metres
4	11°07’25,974’’ 46°10’00,769’’	300 metres	5550 metres
5	11°07’08,273’’4 46°09’44,250’’	300 metres	6015 metres
6	11°05’22,835’’ 46°09’19,305’’	210 metres	7282 metres

The access point WRT-54G supplied by Linksys™ has been selected as basic component for the WLAN point-to-point connection (see Fig.4). The choice of such a model was mainly motivated by the increased degree of flexibility provided by its firmware that is supplied in open source. In particular, we considered the use of DD-WRT firmware [12] that allowed us to set the RF power up to the maximum of 251mW and to implement a scanning function for the 13 available channels. The wireless PCMCIA card selected for our application is the AIRONET 350 supplied by Cisco™. Such a wireless card is supplied with two MCXX connectors able at supporting connections with two external antennas.

Fig.4. The WRT-54G AP and the AIRONET wireless card

The omni-directional antenna employed at the gateway site is shown in Fig.5. It is a slotted waveguide antenna [13] with a 14dB gain and circular radiation pattern.

Fig.5. Slotted waveguide (omni-directional) antenna employed at the gateway

As far as the directional antenna installed at the remote VoIP terminal site is concerned, we considered a parabolic antenna [13] with 21dB gain. The antenna has been physically realized by using a zinc stitch (see Fig.6). In such a way we obtained a lighter antenna that is more resistant against wind effects. In order to conclude this sub-section, we can show in Fig.7 how we realized the remote VoIP site as a "mobile VoIP station" installed in a car. The VoIP terminal is actually a notebook. Note in Fig.7 the parabolic directional antenna employed for VoIP transmission to the gateway. The laptop mounts the AIRONET wireless card (if the conventional IEEE 802.11b/g connection is considered) or it is connected with the WRT-54G access point (if the WDS is employed). Using this kind of mobile station, we could have experimented the quality of the VoIP connection in the different configurations shown in Tab.1.

Fig.6. Parabolic directional antenna installed at the remote site

Fig.7. The mobile VoIP station

B. *Experimental results about Q.o.S.*

The Quality of Service (Q.o.S.) in the considered VoIP system has been measured not in terms of normalized throughput, as usual in IP-based networks, but in terms of the perceived quality of the voice conversation. For this reason, we have shown here experimental results related to the most critical aspects in voice service delivery:

- *Maximum delay*. It is the maximum time delay in the reception of a voice packet [14]. When the delay exceeds a given value, the packet is marked as missed and an audio interruption is caused by this. Maximum delays exceeding 400-500msec are generally regarded with suspect by VoIP providers;
- *Maximum Jitter*. The jitter is defined as a variation in the delay of the received packet. At the sending side, packets are sent in a continuous stream, with packets spaced evenly apart. Due to network congestion, improper queuing or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant [15]. In general, a maximum jitter exceeding 50msec can involve a perceptible degradation in the voice conversation.
- *Perceptual Evaluation of Speech Quality (PESQ)*. It is an objective measurement of the quality of a voice conversation defined by the ITU Recommendation ITU-T P.862 [16]. PESQ methodology compares the original voice signal and the received (distorted) one, using the perceptual biological audio model described in [16]. The PESQ scale ranges from 0.5 (not-audible voice signal) to 4.5 (perfect voice signal without audio impairments).

Our “scaled” experimental setup works in networking conditions that might not be realistically reproduced in the real test-site. In fact, the experimental trials were carried out in an “open” Wi-fi network, with concurrent access to the network performed by other users present in the area. We can say that this scenario could be “futuristic” and “optimistic” for the African zones. In fact, it considers the possibility that, in the next future, the “Wi-fi community” may grow also in developing countries. In any case, the testing scenario with concurrent network access can be regarded as a “worst-case scenario” for testing the proposed VoIP system. The VoIP streaming has been produced by transmitting a conversation of 180 seconds. Among the codecs provided by the SIP protocol, we employed the G.711 one that preserves the quality of the voice signal equal to the original one, avoiding perceptible losses. In such a way, the only measured degradation is related to network impairments.

In Fig.8 we can note that the maximum delay strongly increases with distance if the conventional IEEE 802.11b standard connection between remote site and gateway is considered. On the other hand, the maximum delay consistently decreases if the WDS connection is adopted. Same order of results is shown in Fig.9 about maximum jitter. It is noticeable that maximum jitter remains almost constant for the WDS solution as the connection distance increases. Such results are justified by the fact that IEEE 802.11b/g connection manages the communication between wireless card and AP in concurrent way by using a contention mechanism. On the other hand, WDS manages the communication among different APs in coordinated way by using the Point Coordination Function (PCF) modality. Such results concerning networking aspects substantially impact on the perceived quality of the received voice signal measured by PESQ indicator. In Fig.10, PESQ results are drawn vs. connection distance. Readers can note that PESQ provided by IEEE 802.11b connection fairly decreases with distance (although remaining equal to 3.5 that means good quality). PESQ results provided by WDS connection are almost constant and very close to the excellence threshold of 4.5 (truly they are comprised within 4.3 and 4.5).

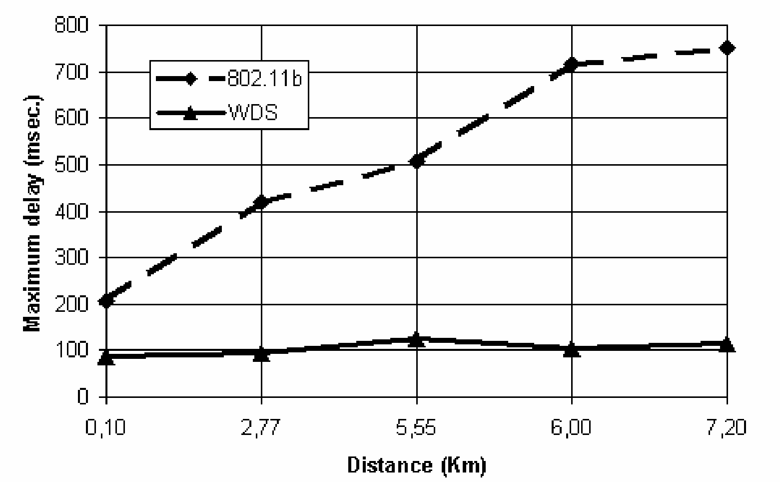

Fig. 8.Maximum delay measured vs. connection distance for the standard 802.11b connection and for the WDS connection

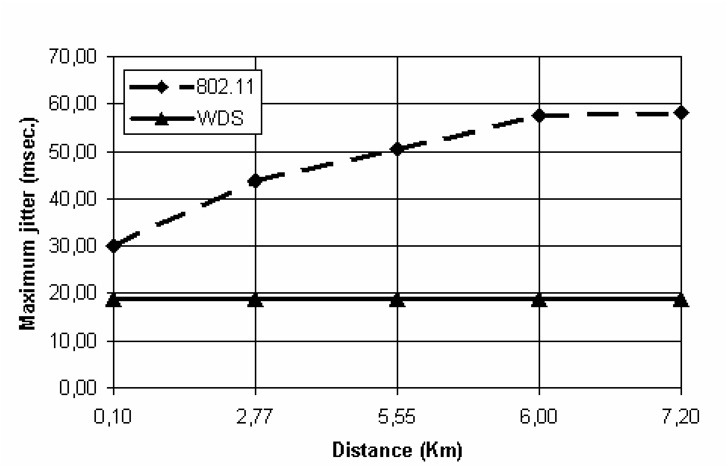

Fig.9.Maximum jitter measured vs. connection distance for the standard 802.11b connection and for the WDS connection

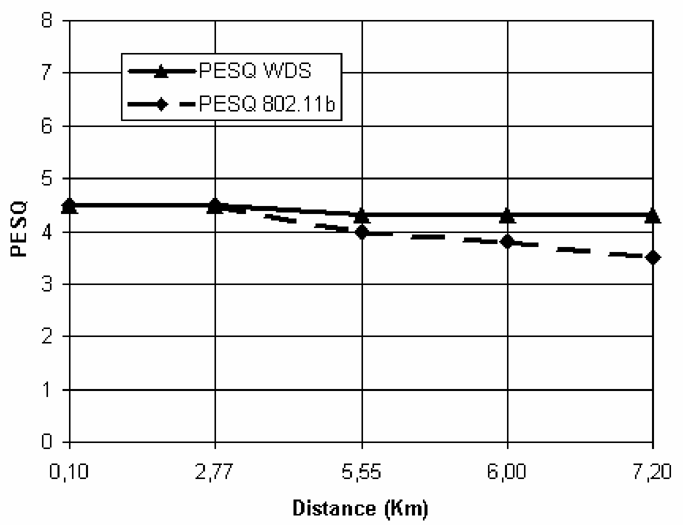

Fig.10. PESQ vs. connection distance measured for the standard 802.11b connection and for the WDS connection

To conclude this section, we report some very short consideration about cost/benefit tradeoff. The total estimated cost for the wireless communication hardware is 324 Euros (about 414 USD). Software is totally open-source and cost-free. The only extra-hardware cost is for the fixed telephone subscription of PSTN connection. It should be said that the ease of installation and use makes the proposed solution very attractive. On the contrary, the installation of a dedicated telephone line would have involved costs of the order of several thousands of USD for interested users. Moreover, the hardware configuration at the gateway site can be arranged by using an old-fashioned PC (a machine with Pentium 133 MHz processor, 16M of RAM and 3.2GB of HD is sufficient to support the ASTERISK server, using Linux Debian 3.0 release). Also the cost of the FXO card is very cheap (about 50 USD).

V. CONCLUSION

In this paper, we presented a full-wireless VoIP system based on Wi-fi radio bridging aimed at interconnecting a remote disconnected site with a gateway served by the PSTN. Such a solution has been explicitly designed for disconnected areas of developing countries not reached by any kind of network infrastructure, but quite close to connected towns. The employment of low-cost license-free Wi-fi transmission and open-source VoIP software tools (SIP and ASTERISK) seem to be key points that can make this solution very attractive in the context of the “Digital Divide” reduction.

VI ACKNOWLEDGEMENTS

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