Abstract—Mobile telephony brings economic and social benefits to its users. As handsets have become more affordable, the ownership has reached staggering numbers, even in the most remote areas of the world. However, network coverage is often lacking in low population densities and low income rural areas of the developing world, where big telecoms often defer from deploying expensive infrastructure. To solve this coverage gap, we propose VillageCell, a low-cost alternative to high-end cell phone networks. VillageCell relies on software defined radios and open-source solutions to provide free local and cheap long-distance communication for remote regions. Our architecture is simple and easy to deploy, yet robust and requires no modification to GSM handsets. The performance evaluation, done by measuring the call quality metrics and the system capacity under a realistic rural-area network load, shows that VillageCell is indeed an attractive solution for rural area voice connectivity.

I. INTRODUCTION

Voice communication is extremely important in rural areas of the developing world. The lack of transportation infrastructure, high illiteracy levels, and frequent seasonal migrations are some of the characteristics of rural areas that emphasise the need for real-time voice communication. In addition, even more than in the developed world, voice communication in the developing world is a strong enabler of political freedoms, economic growth and efficient health care [5], [2].

In our previous work [4] we investigated how rural Africans indigenize voice communication tools and use voice-over-IP (VoIP) applications. Our findings show that, despite having global connectivity, rural dwellers prefer VoIP applications for local, intra-village, communication. While VoIP communication experiences few problems in the developed world where high quality connectivity is available, due to numerous technical obstacles rural area wireless networks cannot successfully carry VoIP calls even within a single village.

Cellphones are another option for voice communication. They are robust low power devices with a very simple and intuitive user interface, which makes them highly suitable for rural areas in the developing world where energy and infrastructure shortages, as well as illiteracy, are usual problems. Indeed, cellphone penetration has skyrocketed in the developing world [1]. Large telecom operators, however, remain reluctant to deploy cellular infrastructure in remote areas. Deployment of cellular networks is complex and requires installation of Base Tranceiver Stations (BTS) and the supporting infrastructure. The installation cost is high, and it remains difficult for the operators to establish a profitable network in areas with low income and population density.

In this paper, we propose a cost effective architecture, dubbed VillageCell, for a GSM cellular network in conjunction with a local rural-area network for VoIP services. The solution uses a Software Defined Radio (SDR) controlled by a software implementation of the GSM protocol stack, called OpenBTS 1, which abstracts the BTS and network components into a single entity. OpenBTS uses SDR for transmitting and receiving in the GSM bands and serves as a local cellphone base station. To extend coverage, through a local wireless network, we interconnect multiple BTSs. One or more Private Branch Exchange (PBX) servers implemented in Asterisk 2 are also in the network and are used call management and routing.

Integrating GSM with VoIP telephony in this manner is cost effective: OpenBTS provides cellular services for a fraction of the cost of a commercial base station, while a local wireless network brings free VoIP-based communication to cellphone users. In summary, VillageCell delivers free local and cheap long distance cell phone communication; it supports short messaging service (SMS), does not require any modification on the end-user devices and works with existing SIM cards.

Implementation: We implement an instance of VillageCell in a lab setting using USRP2 3, a commercial SDR platform that natively supports OpenBTS software and commodity PCs running Linux and the Asterisk software. Connection among the components is established through Linksys WiFi routers.

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1 http://openbts.sourceforge.net
2 www.asterisk.org
3 www.ettus.com
than 3 is considered acceptable.

ranges from perfect (5) to impossible to communicate (1). Any score higher than 3 is considered acceptable.

intraBTS | InterBTS | InterAST
---|---|---
iperf-generated TCP | 1.09% | 1.32% | 1.81%
Trace from Macha, Zambia | 0.69% | 0.82% | 0.88%

<table>
<thead>
<tr>
<th>Backgroun d traffic [Mbps]</th>
<th>InterBTS</th>
<th>InterBTS</th>
<th>InterAST</th>
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</table>
| 0 | 0.4% | 0.6% | 0.8%
| 2 | 0.8% | 1.0% | 1.3%
| 4 | 1.2% | 1.5% | 1.8%
| 6 | 1.6% | 1.9% | 2.2%
| 8 | 2.0% | 2.3% | 2.6%
| 10 | 2.4% | 2.7% | 3.0%
| 12 | 2.8% | 3.1% | 3.4%
| 14 | 3.2% | 3.5% | 3.8%

TABLE I

PACKET LOSS WITH REALISTIC BACKGROUND TRAFFIC.

Three different scenarios of a VillageCell phone call exist depending on the relationship between the call origin/destination and the architecture layout (figure 1). In each of the scenarios we measure packet loss, delay jitter and the number of supported calls. We experiment with different amounts of artificial background traffic as well as with a real-world network trace from Macha, Zambia.

**VillageCell call quality:** In figures 2(a), 2(b), and 2(c) we show end-to-end VoIP packet loss, delay jitter and mean opinion score$^4$ in all scenarios. Both VoIP loss and delay jitter grow linearly with the background traffic. The default GSM voice encoding is G.711, a codec with high loss tolerance, and as long as the packet loss stays below 10% speech communication is possible. In our case loss remains under 2% even with very high background load. To cope with high jitter, VoIP applications often implement receiver-side buffers that store packets for some time (usually less than 100ms) and then sent them out to the decoder in regular intervals. In our setup, the maximum jitter is always below 3ms, thus even a short amount of buffering suffices. Finally, MOS values for each of the scenarios with increasing background traffic remain above 4, implying very good call quality.

Next we investigate VillageCell performance when the voice traffic is mixed with a traffic trace gathered from a wireless network in Macha, Zambia. We replay a randomly selected, ten minute long, snippet of traffic from Macha. We use that traffic in the same way we used the UDP background traffic earlier, and measure the packet loss that a single call experiences in each of the three configurations. To put the results in a perspective, we also use iperf-generated TCP traffic as the background traffic and repeat the experiments. Table I shows that VillageCell looses only a small fraction of packets, thus maintains a good call quality.

**VillageCell capacity:** We evaluate the VillageCell capacity when it comes to multiple simultaneous calls. In our VillageCell implementation we establish a call and incrementally add more calls. Once we have all the calls going, we measure the packet loss rate in each of the calls and calculate the average value. We show the results (figure 3) with both no background UDP traffic and with 1Mbps constant UDP traffic. In all four cases calls experience loss less than 0.3% increase in the packet error rate as we activate all six calls.

**IV. Conclusion**

In this paper we presented VillageCell, a low-cost localized cell phone system for rural areas that solves an important problem of providing localized voice connectivity. In addition, VillageCell offers SMS capability, or data-over-voice solutions such as [3], our system also enables free local data service. In future, we plan to develop applications specifically suited for VillageCell’s unique affordances. Finally, the existence of a community WiFi network and our familiarity with the Internet usage and needs of local population [4], present a solid foundation for our planned work on deploying a full-scale VillageCell deployment in Macha, Zambia.

**References**


$^{4}$Voice call quality is often expressed in mean opinion score (MOS) and ranges from perfect (5) to impossible to communicate (1). Any score higher than 3 is considered acceptable.