Experimental Evaluation of a Software Defined Radio-based Prototype for a Disaster Response Cellular Network

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***Abstract*—In post disaster situations it is vital to restore voice and data communication services quickly. Currently, portable wireless systems are used as a temporary solution. However, these solutions have a lengthy setup, limited coverage, and typically require the use of expensive satellite backhaul. Solutions based on cognitive radio mesh networks have been proposed, to exploit self-configuration and spectrum agility. To evaluate their potential, we build a software-radio-based prototype for a multi-cell network that uses an IEEE 802.11's unlicensed wireless communication band for backhaul, and an open-source GSM stack for access. The prototype provides voice communication services. We evaluate the prototype in an open environment. We demonstrate that under the right conditions, the system can support large numbers of simultaneous calls with acceptable quality. However, when the unlicensed band is heavily used, call quality quickly degrades because of interference on the backhaul link. We conclude that in order to provide acceptable quality of service it is desirable to exploit idle licensed spectrum for backhaul communication between base stations.**

***Index Terms*—Cognitive radio, Disaster response network, GSM, OpenBTS, Rapid deployment, Software defined radio.**

I. Introduction

Large-scale disasters can cause significant infrastructural damage. In particular, existing communication networks may get destroyed, hampering the ability of emergency responders to communicate with each other, and of victims to make contact with rescue services and relatives. A rapidly deployable communication network is required, which must be sufficiently reliable and robust to support mission critical requirements, including quality of service (QoS), mobile user support, and interoperability with existing network infrastructure. It should provide wide-area coverage, and support a variety of different access technologies (WiFi, 2G, 3G, 4G etc.), and it should survive for enough time to allow more stable deployments to be put into place. In this paper we investigate a prototype system for rapid deployment in post-disaster scenarios.

Disaster response networks based on software defined radio (SDR) [1] and cognitive radio (CR) [2] have been proposed [3]. These technologies are designed to sense and observe the available spectrum, and to reconfigure their parameters at runtime. This flexibility would, in theory, allow the network to recognise and respond to whatever access services were being requested, and the spectrum agility would allow communication backhaul to be configured autonomously in real time to adapt to the unknown and dynamic radio environment, thus avoiding the need for time-consuming site surveys, multiple access networks, and manual configuration. Given the demands of disaster response, the backhaul network may have to be multi-hop, involving heterogeneous technologies. Although CR and SDR are promising technologies in a maturing research field, there have been few published practical deployments in real environments, and thus their ability to operate in harsh environments while providing support for legacy user equipment and end-user access services has not yet been properly established. In this paper, we present a prototype for a rapidly deployable disaster response network using software defined radio for voice communication. Our first aim is to establish whether or not the technology can support end-user services, and for this prototype, we fix the access technology to GSM, and we fix the backhaul to use IEEE 802.11 unlicensed bands. We use the OpenBTS software implementation of the GSM protocol stack running on consumer laptops, connected to a Universal Serial Radio Peripheral (USRP) from Ettus Research. We connect two OpenBTS base stations using WiFi connections between the laptops, with one of the laptops running the open source Asterisk PBX to manage the call routing and monitoring for each mobile phone which connects to our network. The Asterisk server is able to provide connectivity to the public switched telephone network or to the Internet. We evaluate the prototype experimentally, under different network conditions, and varying the number of real and simulated voice calls. We demonstrate that the technology can support voice communication, maintaining over 40 calls simultaneously with acceptable jitter , packet loss and mean opinion scores. However, we show that this performance is only obtained when the external radio environment is quiet. In more noisy environments, call quality degrades significantly, allowing fewer than 5 calls simultaneously before jitter and packet loss become unacceptably high. Through analysis of the packet traces, we show that the packet losses are caused by radio interference on the backhaul link. The main contribution of this research lies in demonstrating the limits of using unlicensed bands to support voice communication (GSM-VoIP). The results indicate that true spectrum agility and cognitive radio techniques will be required to support the necessary services.

In the remainder of the paper, we first establish the motivation for exploring software defined radio for disaster response. We then describe our prototype network. We then present the experimental design in full detail. Finally, we present the results of the performance evaluation, and conclude.

II. Background and literature review

The main requirements for disaster response networks (DRNs) cover measures including quality of service, robustness, coverage, deployment speed, interoperability and cost effectiveness, as specified by, for example, SAFECOM [4]. Various rapid response network services are already available, provided by telecommunications vendors, government organisations and charities [3], [5]–[8]. These services are a vital and successful part of the response effort, but they are relatively large scale, take significant time to deploy, and almost all require the use of expensive satellite backhaul. Few current service providers offer interoperability, self-organization, or dynamic spectrum utilization. As discussed in [3], [9] and [10], cognitive radio systems have been proposed as solution techniques for immediate flexible deployment, to provide a working network until more stable solutions can be deployed. Such systems should self-organize, to reduce the need for manual configuration. Further, they should be capable of operating in wide range of frequencies to avoid the need for having prior spectrum information and agreement with governments, and should be able to adapt to unknown and changing environments. Figure 1, shows an example of a disaster scenario in which mobile flexible base stations could be deployed to provide a multi-hop backhaul to connect users requesting different services to the remaining infrastructure. As yet, it has not been demonstrated that cognitive radio or software defined radio with an ad hoc wireless backhaul can support the required user services.

Our prototype system consists of a software defined radio GSM access, with an 802.11 wireless backhaul. Some similar systems have been considered in the literature for other purposes. The VillageLink project [11] [12] has developed a platform to provide GSM services in remote rural locations where there is insufficient cellular coverage. A single Base Transceiver Station (BTS) deployment is presented in [11], [13] and [14], but without backhaul connectivity and thus with limited coverage. The closest system to ours, and part of the inspiration for our work, is described in [15] and [16], in which a local cellular network with multiple BTSs and WiFi backhaul is proposed to provide basic voice and text services, also using OpenBTS, USRP and Asterisk. Those systems were successfully field-tested. However, there are significant differences in the design and operating conditions. For example, a more expensive multiradio router is used in [16], which transmits/receive using different channels. The rural networks are intended for longer term stable deployments, and benefit from site surveys and careful network planning. More importantly, because of the remote deployment, minimal interference is expected on the backhaul links, and so the effect was not studied. VoIP performance analysis for 802.11 is discussed in [17], [18] and [19], using simulations and implementation, but for different service environments, voice codecs and without stretching the backhaul link capacity. For example, in [17] the codec used is G.711, with a data rate of 64 kbps, supporting up to 13 calls.

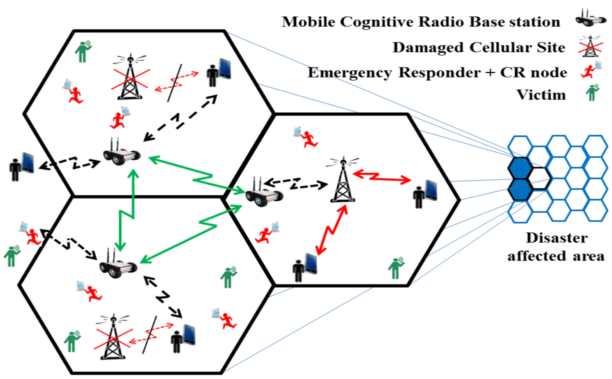


Fig. 1: An example for partially destroyed network extension.

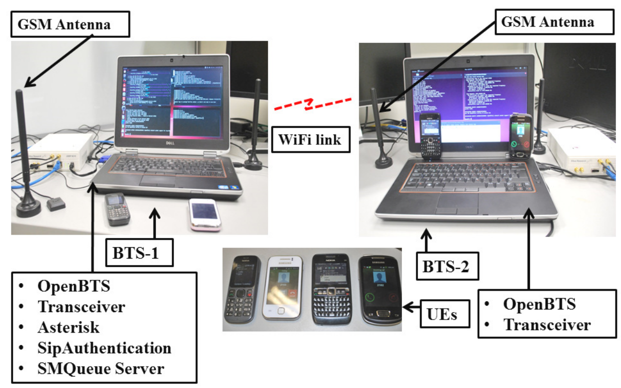


Fig. 2: Prototype of a disaster response network.

III. Prototype multi-BTS GSM-802.11 network

The goal of the research described in this paper is to establish whether or not a software defined radio front end and wireless backhaul could provide the required services of sufficient quality. For that purpose, we focus on enabling a single access technology (GSM), and a simple wireless backhaul using 802.11 on consumer-level peripherals. We choose GSM because of its widespread availability [8], [20]. The software based GSM network and low cost mobile phones, makes it an affordable solution for voice and data services in disaster situations [3]. OpenBTS [21] is so far the only complete open source software based solution for GSM services available. To provide the front end, we use the OpenBTS implementation of the GSM protocol stack, with the radio front end implemented on Ettus Research USRP. USRP is an inexpensive hardware platform for SDRs. It provides the radio communication system by implementing the hardware like filters, mixers etc., in a software on an embedded-system to transmit/receive data.

OpenBTS is a software implementation of GSM protocol stack and replaces core GSM components such as, Base Switching Centre, Mobile Switching Centre, Home Location Register and Visitor Location Registers on a software level. It provides network functions like, GSM registrations, location updating, handover and mobility management. OpenBTS provides "GSM Um" interface to any GSM cell phone within range. It receives GSM signals, demodulates them and converts them to VoIP packets. It can transmit and receive using SDR in the GSM band as a local GSM base station. Where as local wireless network provides VoIP service to cellphone users through existing SIM cards. A call can be established when signalling between the two mobile phones is completed. The signalling is carried out using Private Branch Exchange (PBX).

We use the open source Asterisk PBX implementation, which considers mobile phones as SIP clients through OpenBTS and performs signalling using Session Initiation Protocol (SIP). It works as a server client model and also maintains a database for all the mobile phones associated with it. The Asterisk server manages the call routing and monitoring for each SIP user across the network, and can provide connectivity to the public switched telephone network or Internet. It can be configured on a separate machine or on the same machine of BTS. Other major components of OpenBTS are SIP Authentication server which provides SIP authentication services for registrations and SMQueue server which provides the messaging services to the mobile phones. The backhaul links are implemented as Ad Hoc mode WiFi on the laptops.

IV. Experimental Design

To test the system, as shown in Figure 2, two BTSs are connected over the backhaul using the WiFi link. The two BTSs use separate USRPs. The set up parameters are described in Table I. We test the systems in an open lab environment, and we have no control of the external radio environment. In the lab, there are between 15 and 18 WiFi access points within range. The building opening hours are 7:30 a.m to 10:00 p.m, and most users are active from 9:00 a.m to 12:30 p.m and 1:00 p.m to 7:00 p.m. For the experiments, the two USRPs are placed 8 meters apart, with the GSM antennas 4 meters away from their BTS. Mobile phones are at a distance of between 5 and 25 meters from the antennas. The transmission range for the BTSs is set to approximately 50 meters. We configure one BTS (BTS-1) with an Asterisk server on the same laptop, and BTS-2 on a different machine. The SIP authentication server and SMQueue server are also installed on BTS-1. The experimental scenarios, backhaul link, background traffic and other preliminaries are discussed below.

A. Scenarios

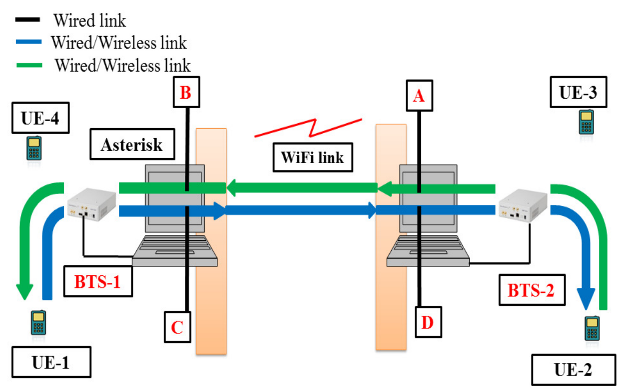
We have fixed two scenarios to test voice quality and backhaul link capacity. The call flows and measurement points are shown in Figure 3.

TABLE I: System parameters of DRN prototype

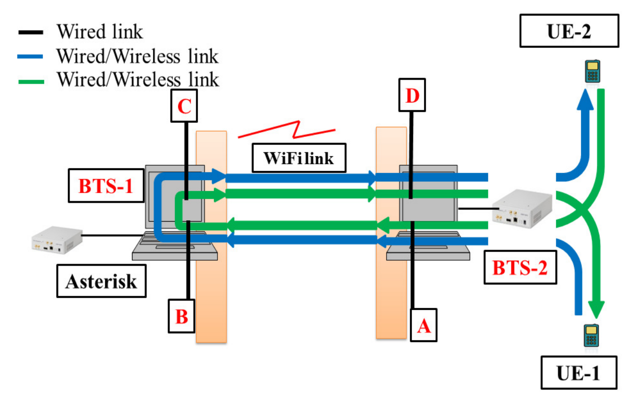
System Parameters	Description
PC	Dell Latitude E6420 (i5, 4GB RAM)
Wireless cards	Intel 1510
Wireless Standard	IEEE 802.11g (54 Mbps)
Voice Codec	GSM 6.10 (13.2 Kbps)
Type of traffic	BTS: SIP, RTP, UDP
Frequency	1800 MHz (GSM) 2412 MHz (WiFi channel 1)
Phones	Samsung S5360 (2), Nokia 101, E63

InterBTS Scenario– The two mobile phones or user equipments (UEs) in a call are subscribed to different BTSs. Both are controlled by PBX-1, which is at BTS-1.

IntraBTS Scenario– Both UEs are subscribed to BTS-2, noting that the Asterisk server is on the machine for BTS-1.



(a) InterBTS



(b) IntraBTS

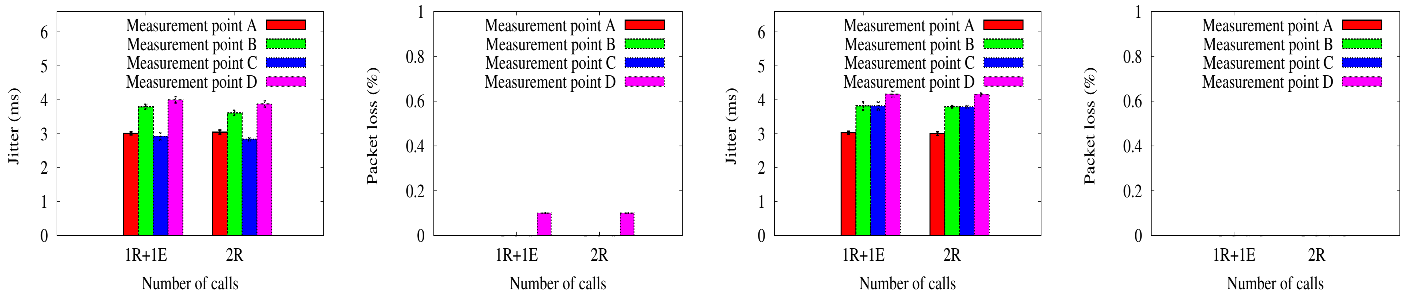
Fig. 3: Call flow and measurement points for DRN prototype.

B. Backhaul Link

For the experiment, we choose channel 1 of 802.11g because it is among the most utilized channels.

C. Background Traffic and Environment

As we conduct our experiments in an open lab, we have no control over external activity on the other networks. Such traffic may cause interference on our backhaul link, and we



(a) InterBTS Jitter (b) InterBTS Packet Loss (c) IntraBTS Jitter (d) IntraBTS Packet Loss

Fig. 4: Equivalence of real and emulated calls in quiet environment.

want to compare the performance of the prototype under different external conditions. We conducted tests at different times over a period of more than a week, focusing on two use cases: (i) *(quiet)* during the early mornings and late evenings, when external user activity was low, and (ii) *(noisy)* during the standard working hours, when external user activity was high. We use a separate laptop which is configured in monitor mode and placed in between both BTSs, to indicate the volume of background traffic. Figure 5d and 6d shows the background traffic monitored in the quiet environment for both scenarios. Similarly, 7d and 8d shows the background traffic monitored in the noisy environment for both scenarios.

D. Measurement points

We captured incoming and outgoing voice traffic at both BTSs using Wireshark and averaged each set of test runs. To understand where, if anywhere, the communication links are degrading, we monitor the traffic at four points. In Figure 3a and 3b, InterBTS and IntraBTS call flow and measurement points are shown and described below.

- A describes the outgoing traffic from BTS-2 for BTS-1.
- B describes the incoming traffic at BTS-1 from BTS-2.
- C describes the outgoing traffic from BTS-1 for BTS-2.
- D describes the incoming traffic at BTS-2 from BTS-1.

E. Equivalent Calls

To increase the number of calls in the system, due to lack of UEs and SIM cards, we have emulated the real calls. The voice call (GSM FR 6.10) codec has a data rate of 13.2 kbps. We have generated and transferred UDP flows at the same data rate using iperf from both BTS-1 and BTS-2. For each additional call, we increased the number of flows accordingly. To validate the emulation, Figure 4 shows jitter and packet loss values for 1 real and 1 equivalent call compared to 2 real calls, for both scenarios. Both show almost identical values, which gives confidence that the results obtained from many emulated calls will correspond to results that would have been obtained from a similar number of real calls. Note here that IntraBTS has twice the flows as InterBTS, because caller and receiver are both on BTS-2. Each RTP frame is of length 87 bytes, which includes 12/8/20 bytes for RTP/UDP/IP header size. The voice packets are sent at regular interval of 20 ms.

F. Performance Metrics

The protocols used are session initiation protocol (SIP) [22], real-time transport protocol (RTP) [23] and RTP control protocol (RTCP) which consists of two streams i.e. upstream and downstream. The following metrics are considered:

Jitter– the mean deviation of the difference in packet arrival time at the receiver compared to the sender and measured in millisecond (ms). It is calculated as [23]

$$J(i) = J(i-1) + \frac{(|D(i-1,i|) - J(i-1)}{16} \quad (1)$$

where D is the *difference in RTP timestamp and RTP time of arrival of a packet*. The division by 16 is to reduce the effects of large random changes.

Packet loss– the number of packets lost compared to number of packets sent, based on RTP sequence numbers.

Throughput– the number of bits successfully received at the receiver per second.

Mean Opinion Score (MOS)– MOS can be used to estimate voice quality as perceived by the users and depends upon the codec being used, which in our case is GSM-FR (6.10). We have used E-model [24] to evaluate MOS, because this model involves codec type and packet loss. The maximum MOS provided by GSM FR is 3.46 for 0% packet loss. MOS is expressed in a range from 1 (worst) to 5 (best). MOS can be calculated from E-Model [24] as

$$MOS = 1 + (R \times 0.035) + R \times (100 - R) \times (R - 60) \times (7 \times 10^{-6}) \quad (2)$$

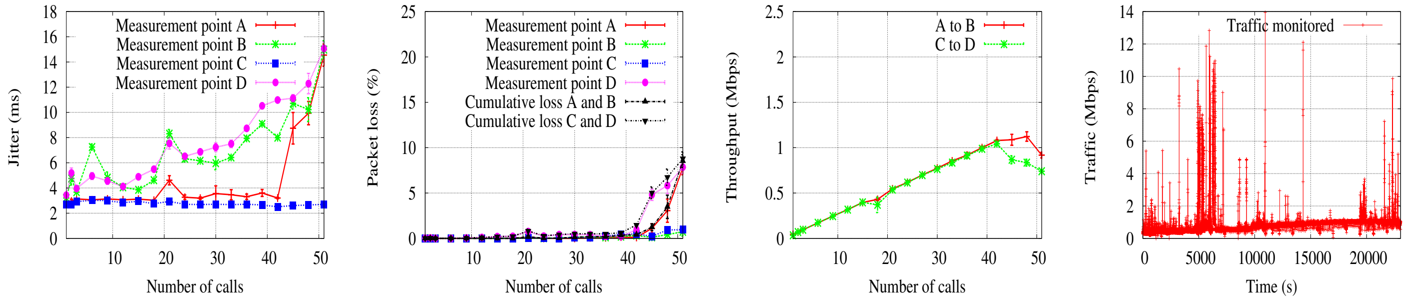
where $\boldsymbol{R}$ is the *rating factor*, which is calculated as a function of the *effective equipment impairment factor*Ie_{eff},

$$R = 93.2 - Ie_{eff} \quad (3)$$

The Ie_{eff} is dependent on packet loss and can be written as

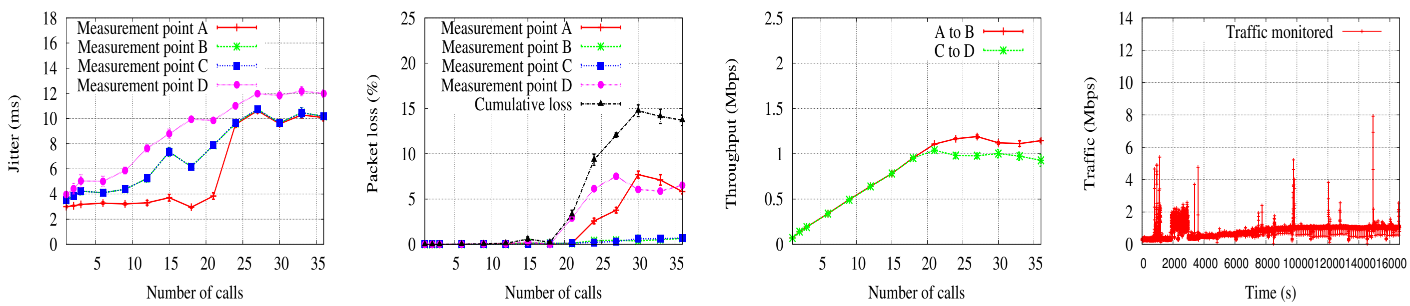
$$Ie_{eff} = Ie + (95 - Ie) \times \frac{ppl}{ppl + bpl} \quad (4)$$

where *Ie* is the impairment factor for 0% packet loss. *bpl* is the packet loss robustness factor and *ppl* is packet loss in percentage for each voice stream. The *Ie* and *bpl* for GSM FR 6.10 are 23 and 46 [25].



(a) Jitter (b) Packet Loss (c) Throughput (d) Background traffic

Fig. 5: InterBTS experiments conducted in quiet environment.



(a) Jitter (b) Packet Loss (c) Throughput (d) Background traffic

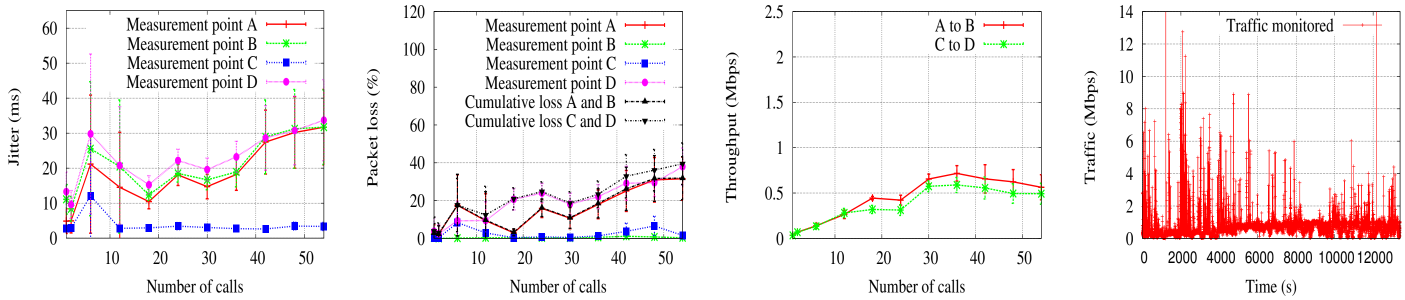
Fig. 6: IntraBTS experiments conducted in quiet environment.

V. Performance Evaluation

To evaluate the capacity of the link, we vary the number of calls (by increasing the number of emulated calls), and measure the impact on jitter, packet loss, throughput and MOS. We run 10 trials of 2 minutes each for every call and take the average of all trials and establish a 95% confidence interval.

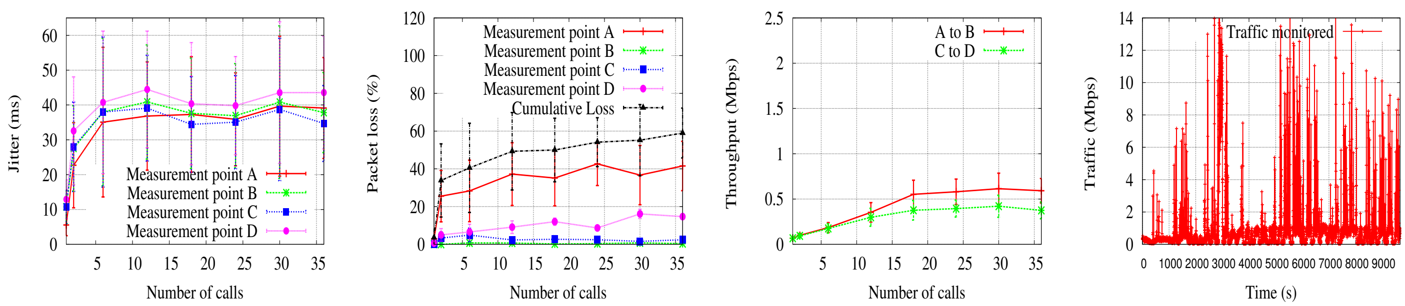
The first set of experiments are conducted in the quiet environment, with results shown in Figure 5 and 6 for InterBTS and IntraBTS cases. Measurement points B and D capture performance statistics for the wireless backhaul link, while points A and C capture statistics for the internal system, i.e., laptops shown in Figure 3. According to Cisco VoIP analysis [26] [27], for acceptable call quality, the average one way jitter should be less than 30 ms. For up to 50 calls, the one-way traffic jitter results are far lower than the 30 ms for both scenarios. Packet loss should ideally be less than 1%, until 5% the voice quality is consider good and until 10% the delayed speech is observed. For the InterBTS case, 50 simultaneous calls can be supported; however, in IntraBTS case, for 25 or more calls, packet loss become too high. This is reflected in the throughput values. As the number of calls increases, the total throughput rises linearly, until close to 50 calls for InterBTS, and until 24 calls for IntraBTS, at which point the communication becomes saturated, and throughput stabilises or declines. Throughout the quite environment experiments, the background traffic remains predominantly below 2 Mbps, with occasional spikes. The sustained increase at 5000s for the InterBTS correspond to the small increases in jitter and packet loss at 20 calls. The MOS for both scenarios can be seen in Figure 9. An MOS below 2.5 MOS is considered to indicate a poor quality call. MOS for the quiet environment is above 3 for the InterBTS scenario until 50 calls, and for the IntraBTS remains above 3 until 24 calls. These results indicate that the combination of a software-defined radio front end and a wireless backhaul is able to support a significant number of GSM voice calls, and thus in principle the technology is feasible for deployment as part of a disaster response solution. Up to 50 calls can be maintained across a single link when one of the UEs is communicating with a front end located on the same machine as the SIP server. When both UEs must share the same wireless backhaul to reach the SIP server, the effective capacity drops by a factor of 2.

The above experiments are conducted when external wireless activity was low. However, in a disaster scenario, we cannot control the external activity, and it is likely that other networks will be establish, or existing equipment may be transmitting. Therefore, we also conduct evaluation when the spectrum is more heavily occupied, as shown in Figure 7 and 8. The increase in the background traffic means increased contention for wireless access. In the InterBTS case, jitter is observed to be approximately 5 times higher than in the quiet environment for even a small number of calls, although



(a) Jitter (b) Packet Loss (c) Throughput (d) Background traffic

Fig. 7: InterBTS experiments conducted in noisy environment.



(a) Jitter (b) Packet Loss (c) Throughput (d) Background traffic

Fig. 8: IntraBTS experiments conducted in noisy environment.

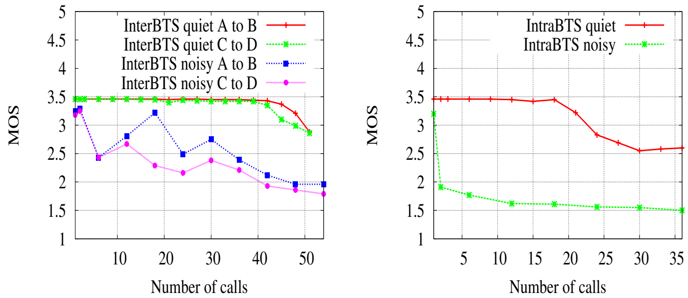
it still remains below the threshold of 30ms until close to 50 calls. However, packet loss rises almost immediately to 20%, and thus very few calls are successful. For the IntraBTS case, jitter rises above 30% almost immediately, and the accumulated packet loss immediately exceeds 20%, and in these experiments acceptable quality was obtained for only a single call. The MOS (Figure 9) for the InterBTS case is noticeably lower than for the quiet environment, and becomes consistently unacceptable after 10 calls. For the IntraBTS case, the MOS value is below 2 for all calls.

We note that the packet loss values are significantly higher at measurement points A and D, and relatively low at points B and C. We believe this is because of the presence of an interfering access point closer to BTS2. From tracing the packets, it appears that when the WiFi transceiver on that laptop listens to the channel, it is frequently unable to find a clear slot to transmit, even though the receiver at the other laptop is relatively interference free. Because of this, the transmitter backs off, and the internal queues begin to fill up, and thus new packets are being dropped in the system before they reach the transmitter. For the reverse direction, the transceiver at C is able to transmit, but collisions are happening at the receiver at point D, thus requiring frequent retransmissions. The effect is magnified at point A in the IntraBTS case because of the doubled flow.

These results demonstrate that the system is not stable in the presence of background interference, and that the system can quickly degrade to the point where it is unable to support more than 1 or 2 simultaneous calls. The performance degradation is due to the wireless backhaul, while the software radio front end is capable of supporting the communication. We speculate that backhaul performance could be partially restored by switching channels. However, in a disaster scenario, we will have no prior knowledge of spectrum use, and we will have little control over other devices in the area which may appear without warning. Thus we believe this indicates that there is a need to utilise cognitive radio techniques, both for the enhanced sensing and spectrum agility, and for the ability to exploit white spaces in licensed spectrum. In particular, if the existing network infrastructure is partially destroyed, then it is likely that licensed spectrum is no longer being used. In addition, we believe that lower frequency bands than those for 802.11 will allow us to extend the range of the backhaul communication.

VI. Conclusion

In disaster response, communication is vital, and yet in many scenarios it is possible that much of the communications infrastructure is destroyed. In these cases, it is critical that a communications network is deployed quickly; this network should offer multiple services to legacy devices, should interoperate with whatever remains of the existing infrastructure, and should provide service until more stable solutions can be deployed. We have designed, implemented and evaluated



(a) InterBTS (b) IntraBTS

Fig. 9: Mean opinion score.

a hybrid software-defined radio access and 802.11 backhaul prototype for rapid deployment to provide cost-effective voice services. We evaluate the prototype in an open environment, and we show that the prototype can support up to 50 simultaneous calls across two base stations. However the wireless backhaul is subject to interference from external transmitters, and we show that in noisier environments, call quality quickly degrades, and the prototype can support only 1 or 2 calls. After a disaster it can be expected that considerable portions of the licensed spectrum will lay idle. We conclude that in order to provide sufficient quality of service when there is serious interference in the unlicensed bands, it is desirable to exploit that idle licensed spectrum for backhaul between base stations, using cognitive radio techniques. As future work, we are now in the process of extending the prototype to use cognitive radios, using TV whitespaces for the evaluation.

ACKNOWLEDGEMENT

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