Implementation of USRP (Universal Software Peripheral Radio) as OpenBTS for Quadruple Play Services

Hafidudin¹, Muhamad Fahru Rizal², Dadan Nur Ramadan³

¹,³Telecommunication Engineering, Faculty of Applied Science Telkom University, Bandung, Indonesia
²Computer Engineering, Faculty of Applied Science Telkom University, Bandung, Indonesia

*hafid@tass.telkomuniversity.ac.id

Abstract. The smallest area of mobile phone service called cell. In one cell area, there is one BTS device. Now the people in the world have a mobile phone, but the people who live in remote areas still poorly understood about how to using it. Therefore, OpenBTS is need for people who do not get the services local mobile operators, as well as in areas of natural disasters. In addition, this OpenBTS it is good to know as a new technology in education. This research proposed OpenBTS using hardware USRP and transceiver antenna for transmitting radio signals at frequencies of GSM 900 MHz. OpenBTS connected to a VoIP service for quadruple play service. All software in this OpenBTS using Linux as operating system, the software is GNU Radio to control the USRP, Open BTS to control the BTS operation, and the IMS Core as server on a VoIP service. From the results of research conducted to measure QoS measurement results show that VoIP networks connected with OpenBTS meet the QoS VoIP standard (Delay = <50ms, Jitter = <15ms and Packet Loss = <1%). Testing of OpenBTS parameters related to transmission using Tems Investigation Software obtained an average level of Rx = -55 dBm, average Rx Quality = 1.17, and average SQI = 13.59. No significant difference in communication between MS and OpenBTS, OpenBTS and VoIP clients, or between VoIP clients.

1. Introduction
Communication technology is now increasingly evolving and evolving, this development is base on the growing computer. Currently computer applications are increasingly vary with various existing applications. This growing computer capability makes it possible to program a software to create a base station. So, the hardware contained in the base station can be replace with open source software on the computer. This open source software can be develop by everyone [1].

USRP (Universal Software Radio Peripheral) with open source software can be a solution to some of the problems facing the mobile telecommunication world. This alternative technology is very useful for building telecommunication networks in remote, rural, rural and disaster areas [2]. Because if conventional BTS is built, the probability of tower success in those areas is very small and the cost required is quite large [3].
With low capital investment, USRP allows users to use phone and SMS features with voice quality and data transfer as well as telecom operators. This USRP can be apply to regular cellular with either active or non-active sim card. However, this is a constraint in terms of licensing considering all GSM frequencies have booked by the operator. [4]

Regardless of the unfinished regulatory matters, this USRP technology as a new technology in the world of education that will be able to give value in the interests of telecommunications which increasingly become essential for human life [5].

OpenBTS is a BTS (Base Transceiver Station) application that runs on a linux platform and is open software. OpenBTS uses a hardware device called USRP (Universal Software Radio Peripheral). It is this device that connects openBTS with standard mobile phone network (GSM)[6],[7]. OpenBTS also uses asterisk software to interconnect with other telephone networks such as PSTN (Public Switched Telephone Network) or other telecommunications operators using VoIP (Voice over IP).[8]

OpenBTS replaces the traditional infrastructure of GSM operators, from the Base Transceiver Station (BTS) to the rear. From the usual traffic forwarded to Mobile Switching Center (MSC), OpenBTS traffic is terminate on the same box by forwarding data to Asterisk PBX via SIP and Voice-over-IP (VoIP). OpenBTS's heart itself is actually a GNU Radio app, serving as a USRP controller. For voice management and voice (voice) used Asterisk application (SIP VoIP protocol). The functionality of Asterisk is similar to the MSC (Mobile Switching Center) device (hardware) on GSM systems. Asterisk is therefore call softswitch because it based on software [8].

---

![OpenBTS Simple Architecture](image1)

**Figure 1** OpenBTS Simple Architecture [9]

A GSM network is built from several functional components that have specific functions and interfaces. In general, GSM networks can be divided into three main parts, namely:

1. Mobile Station/User Equipment
2. Base Station Subsystem
3. Network Subsystem

In each of the main parts of the GSM network is composed of other integrated parts to support its main function. While other networks that can integrate with GSM networks are other cellular networks (PLMN), home telephones (PSTN), ISDN, and internet-based networks as shown in Figure 2.

Universal Software Radio Peripheral (USRP), is one of the tools used to build an OpenBTS, in the USRP itself is divided into 2 parts: Mother Board (Main Board) and Daughter Board (Child Board).
USRP is produce by Ettus Research, the goal of USRP itself is to facilitate the development of cheap radio software. The workings of the USRP is to connect host computers via USB or high-speed Gigabit Ethernet. This connection allows software to control USRP and set up a signal for sending and receiving data. USRP is a hardware that contains high speed Digital Signal Processing (DSP) based radio software that functions as a transceiver (transmitter and receiver) GSM signal. But not just radio signals, USRP can be set to output AM, FM or TV signals, and all the signals are programmed through the software USRP di produksi oleh Ettus Research [11]

Asterisk is an open source software that usually use to build a communication service system or in other words acts as a telephone exchange on a telecommunications network. Asterisk itself provides convenience to users to develop their own phone service with a variety of applications provided to the user. From the definition of open source itself, means that every developer can view and change the existing source code, so this existing applications can be add easily by each developer. Asterisk can also be say as a complete PBX in the form of software, by providing all the features like PBX. Asterisk is able to run on many OS platforms, including Linux, Windows, BSD, Mac, and others. Asterisk can also connect with almost all telephony-based standards, using less expensive hardware as its gateway [12]

GNU radio is a software tool that can be use to create a software-defined radio. Software defined radio is a radio that part signal processing run software. GNU Radio is one of the software that will be use in operating OpenBTS. One of the advantages of GNU Radio is software with open source code and is free software. GNU radio for various things, ranging from GPS data acquisition, DVB transmitter, data communications, wimax design, radar, and even to make a BTS [13][14].

The purpose of the research is to Implement Asterisk using hardware USRP (Universal Software Radio Peripheral) and some open source software as its supporter measure QoS from VoIP Client relationship with MS on USRP such as throughput, jitter variation, packet loss and delay

2. Methodology

OpenBTS (Open Base Transceiver Station) is a software-based GSM network, which allows compatible GSM standard cell phones to make phone calls without using an existing telecommunications network provider. OpenBTS is the world's first open source software based implementation for the GSM industry standard protocol (GSM Protocol Stack). This device was built using the C++ language and released as GNU Affero General Public License (AGPL) version 3 software.

OpenBTS replaces the traditional GSM network operator switching infrastructure subsystem, from the Base Transceiver Station (BTS) in the direction of the uplink. Change the call forwarding traffic mechanism through switching operator's (MSC) mobile center to use Asterisk PBX via SIP and Voice over IP (VoIP). GSM Air Interface (Um) is implemented using software-defined radio (SDR) on Universal Software Radio Peripheral (USRp) devices [15], [16].

In its simplest form, the OpenBTS allows GSM handsets to be use as extensions in the SIP VoIP PBX. OpenBTS is also the foundation for full-scale cellular systems based on VoIP core networks rather than conventional GSM hierarchies. Compared to conventional GSM systems, OpenBTS offers advantages:

1. Minimum network for single cells, operated without integration into existing GSM networks.
2. OpenBTS is a system of all software-, including software-defined radio, with the concept of open source, which means it is free from licenses.
3. The core network that uses OpenBTS for VoIP based is resistant to packet loss, which allows to cross high-quality IP links.

Physical topology of the system to be create is to interconnect between the Portable BTS client with IP Multimedia Subsystem client, so that will be trunking between core BTS Portable with IP Multimedia Sub system core. For more details as in the following picture:

![Figure 3 System Topology](image)

The workings of the system is to build voice communication between the Portable BTS client with IP Multimedia Subsystem client. For the Portable BTS client itself uses GSM communication while for IP Multimedia Subsystem clients themselves use IP-based wifi network. In order to communicate with each other there is a trunking process between cores BTS Portable and IP Multimedia Subsystem. Inside the asterisk there is a configuration to register the Portable BTS client number where for configuration the IMSI client number is changed to the desired number. Likewise on openIMS Core, must register the client first.

Here is a network implementation of the above used devices as follows:

**a. Primary Devices**
1. USRP type B205 Mini
2. A Laptop
   Specifications: NEC Shieldpro Intel core i7, 4 Gb RAM, 160 Gb Hard Drive, Ubuntu Linux 10.10

**b. Supporting Devices**
1. Hardware
   - Switch Switch
   - Kabel UTP UTP cable
   - 3 buah HP with simcard
   - HP Sony Ericsson k800i HP and simcard
   HP Sony Ericsson k800i, useful for walktest is to see the measured OpenBTS parameters, connected to the software tems investigation installed on the laptop. Software
   - X-Lite
A softphone with signaling SIP protocol used as a VoIP client connected to Asterisk server, which is also use as an OpenBTS telephone exchange.

- Wireshark
  This software is useful for displaying packets that pass through the network interface card of the computer along with the time when the packet passes. It is also use to measure QoS on the relationship of VoIP to VoIP, VoIP to OpenBTS, and OpenBTS to VoIP.

- TEMS Investigation
  TEMS is short for Test Mobile System, which is software and Ericsson output device for drive test (done outdoor) or walk test (done indoor).

2. Software
   - Linux Ubuntu 12.04 LTS is used as an operating system
   - GNU Radio 3.4.2, OpenBTS 4.0 as OpenBTS-forming software.
   - Wireshark to display packets that pass through the network interface card of the computer and the time it passes. It is also used to measure QoS on the relationship of VoIP to VoIP, VoIP to OpenBTS.
   - Tems Investigation is Ericsson’s software and device for drive test (outdoor) or walk test (indoor).
   - X-Lite
     A softphone with SIP signaling protocol used as a VoIP client connected to the Asterisk server, which is also use as an OpenBTS telephone exchange.

3. Result and Discussion
   A. Measurement of OpenBTS Parameters
   The purpose of this measurement is to determine the quality of the transmission from the Mobile Station (GSM Mobile) served by OpenBTS when making a telephone connection. Transmission quality parameters analyzed include Rx level, Rx Quality, and SQI. The size of the transmission quality will provide information whether the OpenBTS network that is implement is feasible and meets the KPI value (Key Performance Index) or not. While other parameters are BCCH and ARFCN to find out the frequency used by OpenBTS.

   a. Rx Level
   The Rx level is the reception signal level when the channel is dedicate to the uplink and downlink directions. Rx Level measurement results are display in graphical form can be in the form of Sub-level Rx (dBm) and Full-Rx (dBm) graphics. The difference is if the Full Rx level displays the number of samples from each level of reception when dedicated and idle channels, while Rx level Sub displays
the sample only when dedicated. The following is a plot of the Rx Sub level (dBm) graph, showing the number of samples from each level of acceptance and the cumulative relative frequency (average sample at a level of acceptance level).

![Figure 5 Graph between OpenBTS clients](image1)

![Figure 6 Graph between OpenBTS clients and VoIP](image2)

From the results of the Rx plot, the level shows that the average Rx level of the two scenarios is in the range of -55 dBm. Based on observations when collecting data, the reception level decreases when the distance is more than 20 meters, reaching -80 dBm and the OpenBTS signal will be lost when the distance exceeds 25 meters. Whereas seen from 2 different scenario graphs there is no significant difference between calls between OpenBTS clients and VoIP networks. It can be conclude that the transmission quality of the OpenBTS antenna beam can still be say to be in a good level if the distance does not exceed 25 meters.

b. Rx Quality

Rx Quality is obtained from the transformation of BER (Bit Error Rate) on a scale from 0 to 7. In other words, Rx Quality is the most basic measure that simply describes the average BER with a period above 0.5 seconds. Rx Quality is measure when the dedicated channel is for the uplink and downlink directions. Plotting the Rx Qual Sub chart from the results [enhancement of RX Quality shows the number of samples from each measure of reception quality, as shown in the following graph:

![Figure 7 RX Qual OpenBTS clients](image3)

![Figure 8 RX Qual between OpenBTS clients and VoIP](image4)

Both of the Rx Qual graphs state the specified number of samples in a Rx Qual value and the percentage of the sample at that point for the entire sample. For example in Figure 4.6 for the value of Rx Qual 0 there are approximately 280 samples of the total number of samples, namely 481 and the number of percentages namely (280/481) x 100% ≈ 60%.

c. SQI (Speech Quality Index)

SQI is a sound quality index measured at a dedicated channel for both uplink and downlink directions. This SQI estimates sound quality, clear or not sound quality produced by a cellular network that is
perceive by listeners. The SQI index ranges from -20 to 30. The SQI value is directly proportional to the Rx Qual value. If the reception quality is good (Rx Qual), then the sound quality received by the receiver is also good, and vice versa. The following is a plot of the SQI graph from the measurement results which shows the number of samples from each level of acceptance and the cumulative relative frequency (sample average at a level of acceptance level) generated during dedicated channel conditions.

![SQI OpenBTS clients](image1)

![SQI between OpenBTS clients and VoIP](image2)

**Figure 9** SQI OpenBTS clients  
**Figure 10** SQI between OpenBTS clients and VoIP

The SQI parameter standard of the KPI can be said to be good if it is in the range of 18 to 30. From the graph, it can be seen that more than 90% of the sample has a SQI index of 20 and the distribution is at -8 to 18. From the graph scenario, 1 produces a mean of 15.85 and the scenario graph 2 means 13.59. This SQI is closely related to AMR values. So that it can be concluded that the SQI criteria are good.

### B. QoS (Quality of Service) Measurement

Measurement of Quality of Service to measure parameters that support the quality of VoIP services. Parameters measured include delay, jitter, throughput, and packet loss. The parameter values obtained are compared with the existing QoS standard values, whether they are within predetermined standards or not. While to get the values of QoS parameter itself is used Software Wireshark as network protocol analyzer.

As a good reference or not the values of QoS parameters obtained, then the standards of some institutions serve as a reference, among others:

- Jitter is <50 ms (ITU-T G.1010), and is <30 ms (Cisco)
- Delay is best worth between 0 - 150 ms (ITU-T standard)
- Packet loss <1% (ITU-T standard)

Measurements of QoS are done with varying traffic backgrounds on the network. This is done to test how much the server's ability to serve VoIP calls. Measurements are made by communicating between the VoIP client and OpenBTS. It is done for 30 seconds and involves 5 traffic backgrounds i.e. 0 Mbps, 20 Mbps, 40 Mbps, 60 Mb, and 80 Mbps.

#### a. Delay (Inter arrival Delay)

Delay is the time it takes a packet to move from the sender to the receiver. The unit used is milliseconds (ms). The delay measurement aims to find out how fast the network is use in forwarding packets from the sender to the receiver. The traffic background variation give to know the magnitude of the effect when the packet is send to the end user for its quality degradation.

Measurements made by communicating to VoIP services. During the communication process, the packet capture process done on the client side, with 30-second observation time. The process done by involving the retrieval of data 30 times each scenario so that the average results obtained accurate delay.
In Figure 11 we can see the comparison of delay values from VoIP to openBTS communication with traffic background from 0 mbps up to 80 mbps. From the measurement results, obtained that the delay value. the resulting value is still in very good level according to ITU-T standard.

b. Jitter (Interarrival Jitter)
Jitter is a delay variation that occurs due to instability of network conditions so that the receipt time of packets in the receiver vary. The unit used is milliseconds (ms). The purpose of this measurement is to know the amount of jitter obtained from each scenario. The traffic background variation given to know the magnitude of the effect when the packet sent to the end user for its quality degradation. Jitter is a typical problem in connectionless or packet switch networks. Cisco determined that jitter for realtime communications such as interactive video and voice should not exceed 30 ms.

From Figure 12, it can see jitter measurement result from communication between VoIP to OpenBTS client, this is the same as delay measurement result. So, it can be concluded that the jitter is directly proportional to the delay, the resulting jitter value was categorized very good.

c. Throughput
Throughput is the quantity that shows the number of successful bits sent in the data transmission process of a network, from source to destination, compared to the time range of observation or transmission. The units used can be in bps (bits per second) or bytes per second, but it can also use the scale of multiples (kbps, Mbps, etc).
From figure 13 it can be concluded that the more network traffic due to the intensity of advertisement interval delivery, the less the throughput will be generated. The result of this throughput measurement can we compare with the delay value, where the bigger delay will be the smaller throughput value obtained. This is because when the delay is large, the number of successful packets received will be less and less with a certain time. This proves that the throughput is inversely proportional to the delay.

In addition to delay factors, the influence throughput value are codec and bitrate used during communication process. In the wireshark captured frame size of 214, there are actually additional headers when the packet passes through the cable transmission medium, the header is preamble, Start Frame Delimiter (SFD) and the Cyclic Redundancy Check (CRC). The header counted on frame size on wireshark, but the additional bytes are still consuming bandwidth. Therefore the bandwidth calculation per call must keep adding those bytes. So the total bandwidth per call captured on wireshark will be more than 10,700 bytes / sec.

d. Packet loss

Packet loss is the number of packets wasted during the delivery process compared to the number of packets that survived. Failed delivery of packets due to damage / lost due to something. The unit used is percent (%). Packet loss occurs due to UDP protocol usage in real time communication. This protocol is connectionless, the packet will not sent back in case of a delivery failure. Problems arise if the packet loss is very large. Packet loss for VoIP and interactive video applications can tolerated up to 1% (Cisco).
From the packet loss graph obtained, it can be seen that from the measurement scenarios and variations of background traffic given from 0 Mbps to 80 Mbps still produce packet loss of 0%. This indicates that all RTP packets at the time of communication sent perfectly without missing data, so the communication both goes well. This is evident by looking at the bitrate on the wireshark on the sender side and on the receiving end.

4. Conclusion

Based on the results of the implementation process, testing, and analysis it can be concluded as follows:

1. OpenBTS implementation for Quadruple Play service was successful, proven by successful communication of telephone communication from MS OpenBTS to VoIP client using X-lite softphone on laptop.
2. Testing of OpenBTS parameters related to the transmission analysis by performing a walktest using Software Tems Investigation conducted with two communication scenarios indicates that the OpenBTS network is feasible despite its small reach. The resulting value is the average Rx level of -55 dBm, Rx Quality averages 1.17, and the average SQI is 13.59.
3. QoS measurement results show that VoIP networks connected with OpenBTS meet the QoS VoIP standard (Delay = <50ms, Jitter = <15ms and Packet Loss = <1%).

References

[1] L. C. Tran, D. T. Nguyen, F. Safaei, and P. J. Vial. An experimental study of ofdm in software defined radio systems using gnu platform and usrp2 devices. In 2014 International Conference on Advanced Technologies for Communications (ATC 2014), pages 657–662, Oct 2014.
[2] Khyati Vachhani and Rao Arvind Mallari. Experimental study on wide band FM receiver using GNURadio and RTL-SDR. In 2015 International Conference on Advances in Computing, Communications and Informatics, ICACCI 2015, pages 1810–1814, 2015.
[3] Michael Adeyeye Email author Gardner-Stephen and Paul. The Village Telco project: a reliable and practical wireless mesh telephony infrastructure. EURASIP Journal on Wireless Communications and Networking. (DOI: 10.1186/1687-1499-2011-78), 2011.
[4] Kinjal Aggrawal, Khyati Vachhani, Reconfigurable cellular GSM network using USRP B200 and OpenBTS for disaster-hit regions, IEEE 13th Malaysia International Conference on Communications (MICC), pp 141-146, 2017.
[5] Victor M. Hinostroza Zubía, Héctor García Guzmán, Learning Digital Communications through Software Defined Radio, Número Especial de la Revista Aristas: Investigación Básica y Aplicada. ISSN 2007-9478, Vol.6, Núm. 12.pp 189-193,. Año 2018
[6] K. Sankhe, C. Pradhan, S. Kumar, and G. R. Murthy, “Cost effective restoration of wireless connectivity in disaster hit areas using openbts,” in India Conference (INDICON), 2014 Annual IEEE. IEEE, pp. 1–6, 2014.
[7] J. Mpala and G. van Stam, “Open bts, a GSM experiment in rural zambia,” in e-Infrastructure and e-Services for Developing Countries. Springer, pp. 65–73, 2012.
[8] A. A. Zamzami, E. P. Devara, J. Pramana, A. Sudarsono, and A. Zainudin, “Reliability analysis of gsm network using software defined radiobased system,” in Electronics Symposium (IES), 2015 International. IEEE, pp. 274–279, 2015.
[9] Ray-Guang Cheng, Wei-Lin Hsu & Ping-Chen Lin (2015) A Framework Design for Load-balanced Green Access Networks supporting GSM Femtocell, Smart Science, 3:1, 40–45, DOI: 10.1080/23080477.2015.11665635, 2015.
[10] Bruno Dzogovic, Van Thuan Do, Boning Feng & Thanh van Do, Building virtualized 5G networks using open source software, IEEE Symposium on Computer Applications & Industrial Electronics (ISCAIE), 2018
[11] T. Ulversøy, “Software defined radio: Challenges and opportunities,” Communications Surveys
& Tutorials, IEEE, vol. 12, no. 4, pp. 531–550, 2010.

[12] Z. Li, J. Tang, X. Zhu, and C. Kai, “Simple gsm base station based on universal software radio peripheral,” in Computing, Communication and Networking Technologies (ICCCNT), 2014 International Conference on. IEEE, pp. 1–6, 2014.

[13] Faustine Anthony and Maria Gabriel, Open Source Cellular Technologies for Cost Effective Cellular Connectivity in Rural Areas, In International Journal of Computer Applications (0975 - 8887) Volume 146 - No.15, July 2016, page 29-33, 2016

[14] A. A. Zamzami, E. P. Devara, J. Pramana, A. Sudarsono, and A. Zainudin, “Reliability analysis of gsm network using software defined radio based system,” in Electronics Symposium (IES), 2015 International. IEEE, 2015, pp. 274–279.

[15] T. Ulversoy, “Software defined radio: Challenges and opportunities,” Communications Surveys & Tutorials, IEEE, vol. 12, no. 4, pp. 531–550, 2010

[16] Z. Li, J. Tang, X. Zhu, and C. Kai, “Simple GSM base station based on universal software radio peripheral,” in Computing, Communication and Networking Technologies (ICCCNT), 2014 International Conference on. IEEE, 2014, pp. 1–6.