# Session Initiation Protocol Deployment for Virtual Collaboration In Libraries: A Practical Demonstration

**WADA Ibrahim[1], FURFURI, Ibrahim M. M. Ph.D[2]**

*Jigawa State Institute of Information Technology, Kazaure[1], University Library, Sokoto State University, Sokoto[2]*

*KM3, Kano-Daura Road, PMB 5002, Jigawa State – Nigeria[1], PMB 2134, Along Binin-Kebbi Road, Sokoto State – Nigeria[2]*

E-mail: *wadainfoscience@gmail.com[1], immfurfuri@yahoo.co.uk[2]*

## Abstract

**Purpose:** This study appraises the possibility of deploying Session Initiation Protocol (SIP) as Unified Communication tool to facilitate teamwork among librarians and entrenching a coordinated service provision in libraries.

**Design/Methodology/Approach:** The methodology adopted was descriptive research design which involved field study that attempt an in-depth description of a practical demonstration using SIP technology for a unified communication tool in a modern library setting in Nigeria. Four Open Source Software/technology that were used for the study include Office SIP Server, Jitsi, SIPdroid and Multi-user Video Conference applications. After the installations and configurations, the deployed SIP was tested against signal strength of wireless access point over a range of 50 meters in an open space and 30 meters in an office with two walls demarcations.

**Findings:** From the results of this study, it was discovered that the signal strength of the wireless access point under both parameters (i.e. open and demarcated spaces) has significant impact on the quality of audio, video, and multi-user conference calls. The study further established that the signal strength of the wireless access point is an important determinant for the successful implementation of the SIP.

**Implication:** The libraries in Nigeria should incorporate SIP into their ICT adoptions to facilitate teamwork for optimum collaboration in library services delivery.

**Originality/Value:** The deployment ofSIP by libraries in Nigeria is central to the emerging Smart Library which is an unfolding technology for next generation of library's structural development.

**Keywords:** Session Initiation Protocol (SIP), modern Library, Intercom, Virtual Collaboration, ICT adoption

**Paper Type:** Descriptive field study

## Introduction

Library, as a custodian of information resources, serves as backbone of any academic institution. The roles of librarians in ensuring that library services are efficiently and effectively delivered are crucial for educational activities. To provide such unique supportive services, every library's structure is segmented into units or sections according to distinctive functions, services and routine roles. By this, librarians occupy offices based on their respective responsibilities. Librarians can hardly provide effective services without interactive collaboration. Collaboration among librarians forms an avenue for supports, assistance and exchange of ideas. However, distance between offices possess obstacle to free flow of interactions. Even efforts to hold brief meeting always requires invitation to converge in one location. The emergence of GSM was seen as relief to bridge the gap created by walls of offices. Although GSM proves to be a means of communication for an instance call and messages, it is accompanied with exorbitant call rate and recurrent denial of services. The best solution that can soften the obstacle imposed by office demarcations is the arrival of unified communication technology such as Session Initiation Protocol (SIP). SIP offers many advantages including promoting vertical and horizontal communications, augmenting free flow of ideas, improving staff relations, enthusing cooperation, enhancing teamwork and boosting high productivity among librarians through audio, video, message and multi-user video

conference meeting. Drawing from above, this paper aim to achieve the following objectives:

1. To identify Session Initiation Protocol (SIP) applications for unified communication in libraries,
2. To identify the prerequisites for the SIP deployment in libraries,
3. To test run SIP applications for possible deployment in Libraries, and
4. To find out the challenges that may affect successful use of SIP applications in a library.

## Review of Related Literatures
### Session Initiation Protocol

One of the viable technologies, which permit people across the globe to communicate with little or no recurrent costs, is the Voice Over Internet Protocol (VOIP) (Reed, 2016). VOIP, according to Singh (2008) is a Request for Comments (RFC) emanating from Internet Engineering Task Force (IETF) in 1999. By virtue of the rise of the Internet, Voice over IP communication systems provides much more functionalities that enable the reception of high volume of calls (Bradley 2016). Session Initiation Protocol (SIP) trunking started emerging as a response to demand for free communication and video conference without incurring an unbearable cost. Nowadays, VoIP providers offer services that are tailor-made to suit the needs of an institution like a school, a library, small business as well as large organisation or corporations, within an Intranet, Extranet or Internet (Reed, 2016) domain.

Prokop (2013) described SIP as a form of communication channel that is based on RFC 2543 standard, whose second version was called RFC 3261 after several modifications. SIP is an Intranet, Extranet and Internet telephony calls providing multimedia conferences, instant messages and works with existing Wired and Wireless networks within a defined geographically area or over a long distance via Internet. One of the main advantages of SIP trunking is its ability to integrate everything from internet, data, video, audio voice services, all bundled into a single line.

### Components of Session Initiation Protocol

A typical Session Initiation Protocol (SIP) provides establishment and termination of telephony calls as well as multimedia sessions (Baron, 2005) based on its components such as User Agents Client (UAC) and User Agent Servers (UAS). Hochmuth (2008) and Prokop (2016) concurred that SIP components are meant to facilitate localisation, availability, identification, utilisation, negotiation and interactivity between two or more people. Both the Server Agent Client (SAC) and User Client Agent (UAC) ensure that these components are functionally implementable (Silva, 2017; Reed, 2016). SIP provides unified communication channels that include instant audio, video and messages.

### Unified Communication Technology (UCT)

In today's fast-paced world, organised institution need consistent communication and teamwork in every imaginable way, including mobile phones, personal computers, pagers, personal digital assistants (PDAs), video conferencing, voicemail, instant messaging, and e-mail (SANS, 2016). Due to the power and speed of communication gadgets, UCT is now capable of facilitating the convergence of traditional forms of messaging: telephone, voice mail, fax with newer electronic communications such as email and instant messaging via intranet and the internet (Stallings, 2017). Raulea and Raulea (2015) observed that one of the most commonly used and accepted system for teamwork is Session Initiation Protocol (SIP). Notably, SIP provides an integration of real-time communication services such as Instant Messaging (IM) presence, Internet Protocol Telephony (IP telephony), video and data sharing with non-real-time communication services such as unified messaging (UM) and voice mail (Smith and Houghton, 2007). By the characteristics of SIP as unified communication tool, it supports virtual teaming (Stallings, 2017).

### Virtual Teaming/Collaboration

Session Initiation Protocol (SIP) provides avenue for virtual teaming. Virtual teaming entails interaction and collaboration between and among people using network technology within the confinement of specific work routine (Kinney and Panko, 1996; Goodbody, 2005). Collaboration enshrined by SIP harmonizes the team members and makes them interact virtually (Cascio and Shurygailo, 2003).

Raulea and Raulea (2015) portray virtual teaming as a system that engages group of people in using a form of ICT to communicate with one another more often than in face-to-face meetings. It is a remote method of forming cooperation and interaction based on common ideology, views, responsibilities and goals. One of the unique characteristics of virtual teaming is that the members can work across space, time and organisational boundaries with links strengthened by Local Area Network (LAN), Wide Local Area Network (WLAN) and Internet technologies (Kinney and Panko, 1996; Raulea and Raulea, 2015). Alexander, Wijesinha, and Karne (2010) corroborated that virtual team members could make use of the intranet, extranet and internet network in order to interact and collaborate with one another on tasks and projects even though they may be in different geographical locations and working for different organisations.

*WADA Ibrahim[1], FURFURI, Ibrahim M. M. PhD[2]*

## Methodology:

The methodology adopted was descriptive research design which involved field study that attempt an in-depth description of a practical demonstration using SIP technology for a unified communication system in a modern library setting in Nigeria. Four Open Source Software/technology that were used for the study include OfficeSIP Server, Jitsi, SIPdroid and Multi-user Video Conference applications. After the installations and configurations, the deployed SIP was tested against signal strength of wireless access point over a range of 50 meters in an open space and 30 meters in an office with two walls demarcations.Jigawa State Institute of Information Technology, Kazaure metropolis was mapped out for the exercise. The areas include offices, classrooms, and open space.

## Materials/Tools for the Study

The implementation of SIP requires some important technologies such as presented in the table below. The computer system server provides a platform in which OfficeSIP server is installed. The OfficeSIP server makes it possible to create use profile including call ID and facilitates all interactions and transactions between or amongst users. Each user uses laptop or android phone as workstation for commutation. While Jitsi application is installed in the laptops, the SIPdroid applications are installed in android phones.

The Routing server helps in creating Internet Protocol (IP) and Domain Name Service (DNS) to the Computer server, laptop workstations, android phones and the wireless access points. The IP and DNS allow connection, identification and flow of communications. The wireless access points determine the extent to which the workstations may be able to connect and transmit services. The closer the users to the wireless points, the stronger the signal strength the users will receive, the better the quality of audio, video and multi-user video communication. On the other hand, the farer the users to the wireless access points, the weaker the wireless signal strength, the likelihood is it that the quality of audio, video and multi-user video conference may be low.

| S/NO | Tools | Qty | Version |
|------|-------|-----|---------|
| 1. | Computer System Server (Laptop) | 1 | Core i3, 4G RAM, 320G HDD |
| 2. | Laptops | 5 | Core i3, 2G RAM, 120G HDD |
| 3. | Android Phones | 5 | Android v5+ |
| 4. | OfficeSIP Server application | 1 | V3.5 |
| 5. | Jitsi Clients application | 1 | V2.2 |
| 6. | SIPdroid application | 1 | Latest |
| 7. | Multi User Video Conference | 1 | Latest |
| 8. | Routing Server (MikroTik) | 1 | V.6 License |
| 9. | Wireless Access Points (Dlink and Linksys Router) | 3 | Latest |

## Deployment of Sip Server /Client

Many proprietary and Open Source SIP application exist for deployment. Nevertheless, attention is hereby given to OfficeSIP server, Jitsi and SipDroid applications. The following section deals with the installation, configuration and deployment of the SIP applications.

## OfficeSIP Server installation

OfficeSIP can be downloaded fromwww.officesip.com. The installation takes about seven (7) steps after having downloaded the application. Before the installation processes, Dotnet 3.5 framework application is a requirement that must be installed on the computer with Windows operating system serving as the Server.

1. Right-click on the downloaded application and select **Install**
2. Click on Next to continue
3. Click on the End User License Agreement checklist and click on Next

4. Leave the Destination Folder intact and click on Next
5. On the Ready to install OfficeSIP Server, click on Install
6. Wait and click on Yes to security respond
7. The short cut to the OfficeSIP-Control Panel and OfficeSIP- Web Control Panel will be copied to the desktop of the server to mark the end of the installation processes.

## OfficeSIP Server Configuration

Following the proper installation of the SIP server, configuration becomes the next necessary things to do. The steps for the configuration are presented below.
1. Right-click (double click) the OfficeSIP server control panel on the desktop
2. On the displayed officesip control pane login page, click on **Connect**
3. On the displayed panel, there are four menus such as Home, .csv File, VoIP Providers and Settings. Select the **Settings** menu

4. Change SIP domain name to the choice that fits your library or institution
5. Click on .csv File to create users accounts.
6.

## JITSI Installation

Jitsi is a set of open-source projects that allows building and deployment of secure videoconferencing solutions. At the heart of Jitsi support conferences on the internet and Intranet. As an application, it can be installed on Windows OS, Mac OS and Linux OS.

**Steps for installation of Jitsi on Windows operating system are:**

1. Download Jitsi application from https://jitsi.org/index.php/Main/Download
2. Right click on the downloaded application and select open
3. On the Welcome to the Jitsi Setup Wizard, click on **Next**
4. On the End-User License Agreement, select I accept the terms in the license Agreement and click on **Next**
5. On the Destination Folder page, leave everything intact and click on **Next**
6. On the Additional Tasks page, under various settings, tick Auto Start when computer start and Disable check for update column and click **Next**
7. On the ready to install Jitsi, click on **Install**
8. On completed the Jitsi Setup Wizard, click on **Finish**
9. The messenger automatically loads and offers room for insertion of the account created and provided by the SIP Administrator.

**Steps for Jitsi configuration on Computer device**
After the installation, the following steps are taken to configure the Jitsi messenger. It is better to have setup the SIP server and be on the network before configuring the client devices.

1. Click on start menu and select the Jitsi application
2. Click on **Tool** menu and select **Options**
3. Click on Add
4. In the Network column, select SIP network
5. Under User name and password, insert the sip address @ domain and Password provided by the SIP administrator
6. Click on **Advance**
7. Click on **Connection** menu

8. Under the proxy options, deselect configure proxy automatically
9. In the proxy column, insert the IP address of the SIP server
10. In the Port column, insert 5060
11. Under preferred transport, select UDP or TCP
12. Leave every other thins intact and click on **Next** and then **Sign-in**

## SIPdroip Installation

SIPdroid is an Application Package Kit (apk) programme designed specifically for communication on Session Initiation Protocol. It has some basic feature that enable answering calls, making calls as well as calls transfer.

**Steps for SipDroid Installation on Android device**

To install SIP messenger on an Android phone, here are the steps:
1. Download Sipdroid from Google play store at https://play.google.com/store/
2. Accept the terms of License Agreement and click on **Continue**
3. Tap on open and configure the SIP account with the account provided by the SIP Administrator.

**Steps for Sipdroid configuration on Android device**

1. Tap on the installed Sipdroid application
2. Tap on the star-like logo
3. Tap on the SIP account Line 1
4. Under Authorization Username, type in the username provided
5. Insert the password under the password column
6. Under the server or proxy, type the IP address of the SIP server
7. Type the Domain name under the domain column
8. Select use WLAN
9. Tap navigate back
10. Tap the wireless among other options
11. Select Control Wi-Fi Power,
12. Tap on back navigation two times
13. To call, tap on Phone Number column and type the client unique number provided by the SIP administrator

## User Profile

The table 1.1 below presents user profile for the demonstration. The user profile is created in OfficeSIP server application.

**Table 1.1:** User Profile and Caller ID

| S/NO | Username | Caller ID | Password | Email |
|------|----------|-----------|----------|-------|
| 1. | User one | 0601 | 01 | 0601@wada.com |
| 2. | User two | 0602 | 02 | 0602@wada.com |
| 3. | User three | 0603 | 03 | 0603@wada.com |
| 4. | User four | 0604 | 04 | 0604@wada.com |
| 5. | User five | 0605 | 05 | 0605@wada.com |

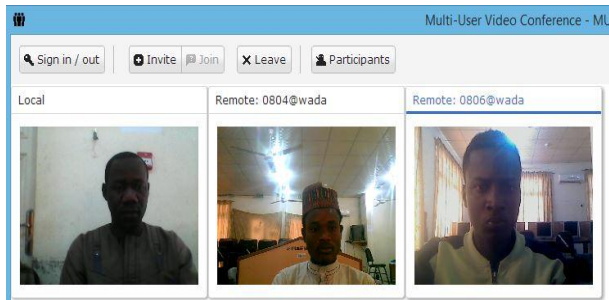*WADA Ibrahim[1], FURFURI, Ibrahim M. M. PhD[2]*

Fig. 1.1: Three People in Conference Call



Fig. 1.2: Two people in Video call

## Direction of Interaction in library with SIP

Librarians perform series of functions by interacting with one another on daily basis. Such interactions require constant information gathering, classifying, storing and sharing, group supporting, meetings, brainstorming and decision-making. The Fig.1.3 below displays the various directions of interaction amongst librarians in a typical library. The ability of SIP to support unified communication by integrating various communication channels has made it worthwhile for library's implementation.
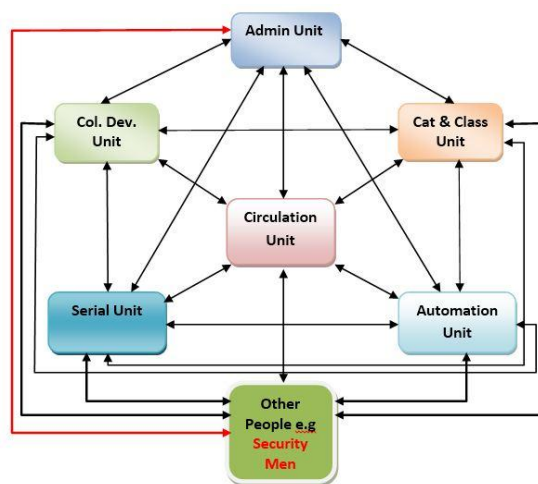


Fig. 1.3: Direction of Interaction with SIP

### Benefits of SIP in Libraries

Session Initiation Protocol offer many advantages to institution, especially libraries in this information technology driven society today. Some benefits that librarians could gain when SIP is deployed in any library is here presented below.

### Information sharing

At one time or the other, librarians need to exchange information. SIP can facilitate easy and fast flow of information. Through any of the multimedia channel, information could easily be transmitted to the other personnel within the library at no cost and without stress.

### Information gathering

SIP is good for collection of polls, distribute questionnaire and gather suggestions by librarians. Gathering of ideas from colleagues and staff members take just a few seconds through the use of Unified Communication System.

### Group supporting

Organised assistance is necessary between and among librarians, as they tend to cooperate for service rendering. The sense of supporting one another with the use of SIP through conference calls and video communication can yield high quality services than individualistic approach to

performing functions. Group of staff in one unit in the library can reach out to other group in other

**Chat and meeting**

Chatting as well as meeting online among staff is now a common routine since the emergence of remote communication through chat messengers, social media, emails, GSM communication and Internet calls facilities. However, SIP supports and facilitates chatting and meeting seamlessly and can be initiated at will by librarians.

**Information broadcasting**

Broadcasting information at a click in one feature of SIP, which should not be undermined. It has ability to facilitate sending information to various people on the network at the same time. This form of reaching to colleagues or coworkers connected with networks and supported by UC has far reaching positive effect on information sharing trends. This is more of alerting services as part of information sharing strategy.

**Idea poll**

There are times when the need do arise to seek people's views or opinions on a matter that affect plans for actions. SIP is a very good platform with which to conduct surveys. Having staff being on a network, request for individual's respective point of views could easily be collected effortlessly.

**Brainstorming through group Decision-making**

SIP supports and facilitates brainstorming. Brainstorming as a decision-making channel is used to encourage sharing and choosing the best ideas and direction to achieving a goal. Brainstorming is a process of thinking together, which offers technique by which efforts are collated to find a solution to a specific problem by gathering a list of ideas spontaneously contributed by members. SIP provides a quick means for tapping the creativity of a limited number of people for gathering large and qualitative ideas.

**Conclusion**

Deployment of SIP is good for interaction and collaboration in libraries. With the available Information Technology facilities such as Local Area Connection (LAN) and Wireless Access Points or Wireless Fidelity (wifi), SIP

sections of the library for unified roles performing.

implementation in libraries can facilitate virtual teamwork. SIP can be the best technology to facilitate unified communication route, given its supports audio call, video call, instant messaging and multi-user conferencing. In order to optimize library services and improve clienteles' satisfaction, SIP can help in promoting vertical and horizontal communications, augment free flow of ideas, improve staff relations, enthuse cooperation, enhance teamwork and boost high productivity among librarians. More visible advantages of implementing SIP in libraries are as follows:

1. It will enable librarians to cut cost on communication as they collaborate remotely
2. It will provide privacy of communication among librarians as they retain the ownership of their conversations
3. It will provide easy and flexible collaboration between and among librarians
4. It will help in enhancing library service delivery as librarians cooperate to tackles dire needs of clienteles through collaborations
5. It will empower librarians to provide responsive answers to the questions of library clienteles
6. Exchange of questions and answers between and among librarians can be prompt, fast and deliberate
7. Free flow of communication either in audio or video calls will boost librarians morale to assist one another as quickly as possible
8. It will facilitate quick and decisive meetings without the physical presence of librarians and stakeholders under one room or hall.
9. It will register librarians' online presence, which also means office presence, thereby providing room for checks and balances among staff of the libraries.

**Recommendation**

The implementation of SIP does not attract recurrent cost of installation and deployment. Moreover, with most institution having installed network facilities such as LAN and WAN for Intranet, Extranet and Internet services and Android phones and laptops now available for many people, the deployment of SIP is more easier than could be envisaged. The accumulating benefits of SIP implementation within shortest and over a long time period cannot be underestimated. Better still, SIP is central to the

*WADA Ibrahim[1], FURFURI, Ibrahim M. M. PhD[2]*

emerging Smart Library—being an unfolding technology for next generation of library's structural development. It is worth having!

1. Libraries wishing to deploy SIP technology to establish unified communication system should use MicroTik and Ubiquity routers because their signal strengths are higher when compared with other routers.

2. One class of IP could be used for connection and troubleshooting. However, port forwarding becomes necessary when different classes of IP are established.

3. Libraries should acquire and install enough access points to boost signal strength in order to avert poor voice and video communication.

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