Name of organization supported	One Destination Center
Project title	VoIP as a model applicable to developing countries
Dates covered by this report	1 August 2009 – 31 January 2010
Country where the project has been implemented	Indonesia
Project leader name	Anton Raharja
Team members (list)	Onno W. Purbo, Nurlina Purbo, Protus Tanuhandaru, Dadan Hatomi, Ashoka Wardana
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Synthesis

Many communities in developing economies do not have access to communications infrastructure. In these cases alternative means of communication, such as Internet Protocol Private Branch eXchange (IP PBX) or softswitch, may be employed. These infrastructure-free alternatives can play a central role in helping these communities to gain educational, employment, and economic opportunities otherwise only afforded by conventional telephony systems, such as Public Switch Telephone Network (PSTN) or Cellular.

IP PBX is a telephone system designed to deliver voice or video over a data network. Just like a normal Private Branch eXchange (PBX) or the conventional phone exchange, IP PBX also connects a number of phone lines (extensions). IP PBX may have benefits over conventional PBX, as it is based on Internet Protocol. IP PBX and may be wireless, can accommodate more extensions than conventional PBX, and enables communication at a rate that is far less costly than long distance and international calling.

IP PBX can also be interconnected to PSTN, allowing users registered to an IP PBX system to dial and receive calls from PSTN or Cellular to Voice over Internet Protocol (VoIP) and vice versa. However, in reality, as in the context of Indonesia, such interconnectivity is limited to one-way calling. This means that a person can only use VoIP to dial PSTN lines or cellular but not vice versa. This is because two-way interconnectivity is forbidden by regulation that governs only outgoing calls from VoIP providers to PSTN or cellular operators. This implies that the government does not fully integrate their numbers into the e164.arpa, the Electronic Number Mapping System association acknowledged by the International Telecommunication Union.

Given the above situation in Indonesia, this project's objective is to enable those deprived of communication, and facing exorbitant telecommunication services, to build their own low-cost communication infrastructure, by providing them with a tested softswitch equipped with enum capability. Enum can translate IP addresses into PSTN or cellular numbers. Once the softswitch is tested, it will be used to establish enum.voiprakyat.or.id to connect VoIP Rakyat, Indonesia's free VoIP provider, to www.e164.org, both of which are public enum directories that can be reached by anyone anywhere, as long as there is Internet connectivity. Through these directories, VoIP users have more options to make and receive calls from and by numbers facilitated by those directories.

This project employs the Briker softswitch, which was built by the project leader, based on a number of components: Free PBX 2.4, Asterisk 1.4, Asterisk2Billing 1.3, and Webmin, all of which are bundled into a linux software that runs on Ubuntu Platform.

However, both the Briker and enum development are subject to a number of problems. First, the modifications required for the softswitch might not perform well under the Session Initiation Protocol (SIP) environment. Even if it does work in these conditions, the softswitch has yet to demonstrate that it can deliver acceptable quality of VoIP calls for a large number of concurrent calls accommodated by the softswitch.

The methodological approach to address the reliability of the softswitch includes conducting a test (1) on whether the Briker, with an additional modification, will run properly in the SIP environment, by sufficiently showing two nomenclatures of VoIP call status: Ack and SIP-bye for successful connected calls and termination respectively; and (2) on the quality of calls the Briker can facilitate, which is usually measured in terms of Quality of Service (QoS) parameters, including delay, jitter, and Mean Opinion Score (MOS). Call quality is mainly determined by codecs, a process that converts analog code into digital code. Different codecs have different compression ratios, which in turn result in different bandwidth requirements and Quality of Service (QoS) in terms of jitter, delay and Mean Opinion Score (MOS) values. Essentially, higher compression rates require more resources to process the calls, which will subsequently require lower bandwidth.

To run the test, we utilized a software called VQManager, a web-based monitoring tool capable of monitoring VoIP quality calls in terms of Quality of Service (QoS) parameters. All features of the monitoring software, including its capability to support an unlimited number of IP Phones, are valid for 30 days. After the trial period expires, the software downgrades to the free edition, which is valid for an unlimited period, but is only capable of supporting a maximum of 10 IP Phones.

Prior to the commencement of this project, activities undertaken went as far as building and using the without understanding the dynamism of the call quality delivered by the softswitch.

Development Problem

As stated earlier, both the Briker and enum development are subject to a number of problems. First, the required softswitch modification may affect performance in the SIP environment. Second, the softswitch has yet to demonstrate that it can deliver acceptable quality of VoIP calls for a large number of concurrent calls accommodated by the softswitch.

Project Process

Two processes that were observed to be successful during the first 6 months of activities:

1. ODC and VoIP Rakyat have established enum.voiprakyat.or.id, a system that verifies the eligibility of PSTN and cellular numbers to be registered to e164.org.
2. Test on Briker Softswitch was successful to the extent that the softswitch has been shown to work in the SIP environment, with the test results attached as Annex 1
3. A manual on VoIP is now available in English http://125.160.17.21/speedyorari/index.php?dir=ebook-voip as well as in Wiki http://opensource.telkomspeedy.com/wiki/index.php/Internet_Telepon and http://opensource.telkomspeedy.com/wiki/index.php/VoIP_Cookbook:_Building_your_own_Telecommunication_Infrastructure

Two processes that did not work as intended, and the reasons why:

1. The SIP firewall penetration test could not be carried out, as the equipment needed to simulate the test configuration of all types of NAT are not available. Such a test, although later suggested by the ISIF committee, is not covered by the budget proposed to ISIF. We suggest that VoIP SIP users who have their access limited by NAT/Firewall should attempt to open the following ports: UDP 3478, 3479 (STUN), UDP 5060 (SIP signaling), UDP 8000-10000 (SIP RTP, data voice, and video) for SIP.
2. The research, although successful in indicating that the softswitch works properly under the SIP environment and in finding out which codec is suitable for the goal of the project, failed to measure the call quality for a number of concurrent calls larger than several endpoints set up for the test. As stated in the research document, this failure is attributable to the technical problems encountered during the test. The monitoring software is reliable to an extent that it is used within a very small size of network connected in LAN topology.
3. The work on building a VoIP cost-benefit analysis tool, as suggested in the previous report, never took place.

Fulfillment Of Objectives

In accordance with both the objectives outlined in the Memorandum of Understanding (MoU) and the abstracts contained in the proposal, the project's objective is to enable communities deprived of a means of communication to build their own low-cost communication infrastructure, by providing them with the softswitch Briker, an Internet Protocol Private Branch eXchange (IP PBX) software partly built on Asterisk and equipped with enum compatibility. This objective has been accomplished to the extent that the softswitch was found to working properly in the SIP environment, that it can be distributed to the community, and that the codec suitable for the goal of the project was identified. Also available as a result of the project is e164.or.id.

The specific objective of turning as many beneficiaries as possible on to VoIP and as a result making them less dependent on GSM, CDMA, or PSTN, is contingent on the upcoming months of the project.

Project design and implementation

During March - April 2009, the team improved the backend and frontend of VoIP Rakyat, which is the free VoIP provider servicing more than 70,000 registered users.

During April - May 2009, team worked to establish enum.voiprakyat.or.id and www.e164.or.id, a system that verifies the eligibility of PSTN and cellular numbers to be registered to e164.org, by validating these numbers to ensure the number being registered is real. Users willing to have their numbers registered to e164.or.id are provided with a Personal Identification Number (PIN), which is used to validate the number(s) being registered, either through a Short Message Service (SMS) or phone call, depending on whether the number originates from PSTN or cellular. Once this block of validated numbers is forwarded to e164.org, a verification process determines whether the numbers already exist within the e164.org database. Any number found to be similar to those in e164.org would be omitted.

During May - June 2009, the team carried out the test on the softswitch in order to determine:

1. Which codec produces acceptable call quality under the constrained circumstances of equipment and bandwidth availability. In particular, the codecs have to be available free of charge and supported by softswitch and softphones. The softphones used included X-Lite, a softphone featured with voice, video, and instant messaging. When any codec in the softswitch matched a similar codec of the softphone, it eliminates the need for transcoding. Transcoding is the process that translates one codec into another, which consumes more CPU resources, thereby degrading the quality of VoIP calls.

2. Whether the softswitch would function well in the SIP environment.

In the future, ODC and ITMN will bring more users in by showing the results of an in-depth evaluation, on cost savings and activities enhanced by IP PBX implementation. ODC will develop a cost-benefit analysis tool to measure how much the same calls would cost if they are dialed using VoIP instead of PSTN or cellular, and the resulting potential cost saving against the equipment purchased, operation and maintenance costs, and the bandwidth used by the system. The tool will be based on any known available standard adopted or built by telecommunication business engagements providing VoIP service.

To determine whether the Briker is in compliance with SIP, we observed that two endpoints in the test are successful in establishing and terminating calls. This is shown by the two states in a call flow status: ACK for successful call and SIP-BYE for terminating calls. We measured the quality of the calls in terms of

Mean Opinion Score (MOS), on a scale ranging from 1 for unacceptable to 5 for excellent.

During the test, we encountered some problems. Two of the tested codecs, iLBC and GSM, generated significant delay, which is unusual given that the test only involved few endpoints which are connected by LAN topology and did not consume large bandwidth. The amount of resources used by the monitoring application, as claimed by VQManager manual, was infinitesimal, or too small to affect the performance of the test.

Despite testing repetition, the problems persisted. In the test result, we have provided some possible answers to why the simulation failed. Furthermore, the ambiguous results pertaining to the call quality have been treated as an anomaly. The values in question were used to generate the MOS values, which we then used to derive our conclusion results. The delay, however, seemed not to have affected the quality of the sound, since the voice transmitted was still audible by the testers.

From the simulation results, we drew the following conclusions:

1. The analysis has shown that of the two unlicensed codecs, iLBC is recommended for its high MOS value, despite its enormous delay as indicated by the simulation. We compromised such delay in favor of MOS values obtained for the codec.
2. Although we have obtained what we consider to be a suitable codec for the goal of the project, it is too early to conclude that the same or better QoS values will be achieved for the same codec for a large number of concurrent calls. If the test had been conclusive without irregularities, we could have used the software to measure the aggregate QoS Metrics of VoIP Rakyat system instead of using the number of endpoints.
3. Irrespective of the result values, we have shown that the application functions well under the SIP environment.

From November 2009 – January 2010, the team has been writing a VoIP manual. Numbering 235 pages, the book—tentatively titled VoIP Cookbook: How to Build your own Telecommunication Infrastructure—contains step-by-step illustrated instructions on how to become a VoIP user and operator. The book is now available at:

- http://opensource.telkomspeedy.com/speedyorari/index.php?dir=ebook-voip

A VoIP wiki has also been written and can be accessed at:

- http://opensource.telkomspeedy.com/wiki/index.php/VoIP_Cookbook:_Building_your_own_Telecommunication_Infrastructure
- http://opensource.telkomspeedy.com/wiki/index.php/Internet_Telepon

Project outputs and dissemination

The project output is a tested, enhanced softswitch with enum capability. The software is bundled in a package available for download from www.briker.org. In addition to the softswitch, there is also an installation and operation manual for users. This manual, available in Indonesian and English, can be downloaded from VR's website:

http://voiprakyat.or.id/pub/manual-briker/Manual_Briker_IPPBX_Administration_en.pdf

Research results pertaining to the softswitch are attached as Annex 1 in this report. The information on the softswitch availability will be disseminated through mailing lists and events, such as workshops and seminars.

As one of objectives of this project aims at integrating VoIP with the PSTN network provided by the national company Telkom, an opinion-editorial piece on this integration was published.[1] In addition to this op-ed, an article featuring the Briker softswitch was also published, with the aim of getting the wider public to understand that they can easily establish a low-cost means of communication.[2]

In closing the project, we will also publish a book on VoIP capacity building. The manual will emphasize enabling the readers to easily install and operate VoIP system, through step-by-step illustrated instructions. Once completed, the book will be distributed in PDF format, available for download from www.odc-foundation.org under a Creative Common License.

The distribution of the book may also be supported by Wiki, a collection of web pages designed to enable anyone who accesses it to contribute or modify content. Through wiki, readers will be able to participate in online discussion and group editing, making the content as dynamic and up-to-date as possible. This may help us reach a wider audience, particularly the marginalized groups without effective digital access.

The Important URLs related project outputs are:

- http://opensource.telkomspeedy.com/speedyorari/index.php?dir=ebook-voip
- http://opensource.telkomspeedy.com/wiki/index.php/Internet_Telepon (Indonesian Language)
- http://opensource.telkomspeedy.com/wiki/index.php/VoIP_Cookbook:_Building_your_own_Telecommunication_Infrastructure.
- http://www.briker.org (SIP Softswitch).
- http://enum.voiprakyat.or.id and http://www.e164.or.id (ENUM Server).

Capacity Building

As a result of the initiation of this project, new equipment such as servers and notebooks are available. These additional servers enable VoIP Rakyat to presumably support more than 700 registered online users and 400 concurrent calls. Prior to the implementation of the project, VoIP Rakyat could accommodate only less than 300 online users and 120 concurrent calls. The two programmers have enhanced the Briker softswitch, as well as the front and back ends of VoIP Rakyat

The project team's members, particularly the engineers and the project leader, have developed their research skills and are now capable of using the proper methodology for measuring VoIP quality. Utilizing software designed to measure VoIP quality enhanced their current skills. What they have achieved is relatively remarkable, considering that no one but the senior consultant of the project is academically qualified to do the research. Likewise, the team's collective administrative skills also improved to an extent that the team is now capable of managing a variety of documentations.

Several VoIP workshops have been given in collaboration with various universities and communities. Planned upcoming workshops are to be held at the Asian Institute of Technology. The more recent workshops partially funded by the ISIF Grant are:

- 13 March 2010: STIKI Malang, Audience 150 persons. Detailed address: Kampus STIKI, Jl.Tidar Raya 100, Malang, Jawa Timur, Indonesia. Telp:(0341)560823, Fax :(0341)562525, Email: stiki@stiki.ac.id, Web www.stiki.ac.id.

[1] Please refer to Annex 6

[2] Please look refer to Annex 2

- 17 March 2010: SMK YASTI, Audience 300 persons. Detailed Address: SMK YASTI, Jalan Veteran No 66, Cisaat, Sukabumi, Jawa Barat, Indonesia 43152, Telepon/fax : (0266) 228203, 230677, Email : iwank66@yahoo.com, Web: http://smkyasti.sch.id/

Project Management

Consultants Protus Tanuhandaru and Nurlina Purbo have overseen the administration of the project Protus assisted the project leader, Anton Raharia, in scientific and technical management. Whenever we encountered technical challenges, the senior consultant Onno W. Purbo provided input as well.

As of the submission of this final report, there is no change to the project management to be incorporated for the duration of the project life.

Project Sustainability

ODC's major objective is provide education and awareness on VoIP technology as low-cost mode of communication. A significant request for VoIP workshops has been received from various communities as well as Interlabs at the Asian Institute of Technology.

ODC is partnering with InfoTech Media Nusantara (ITMN) to sustain the inputs and processes of this project for the future. ITMN is a company that provides IP PBX solutions by using Briker as a softswitch. This collaboration is based on the consideration that ITMN has expertise in VoIP, and this collaboration will promote our project so that it is sustainable and scalable both to a national and international level Some of the ODC project team members, including the project leader and programmers, are erfrom ITMN.

Since the outset of this collaboration, ODC and ITMN have been seeking project opportunities. In March 2009, ODC and ITMN proposed to the Indonesian Ministry of Research and Technology to extend the VoIP Rakyat network. The long-term objective of the proposal is to enlarge VoIP Rakyat's network, by disseminating information about VoIP Rakyat through the ministry network covering the nation. The specific objective proposed to achieve this goal is to increase the communication efficiency of the ministry, by establishing IP PBX connectivity between the research facility situated in Serpong, West Java, and the ministry headquarters located in Jakarta, the capital of Indonesia.[3]

Subsequently, in late April 2009, ITMN and ODC were requested by Axis, a 2G/3G provider in Indonesia, to help establish Axis's VoIP community, by adding VoIP as a feature to one of Axis's products, the 3G service. The proposition to Axis includes establishing interconnectivity between the Axis network and VoIP Rakyat's servers, increasing the capacity for VR's online users, concurrent calls, and handset softphone development. This will in turn increase the capacity of Axis's employees on VoIP. Part of this initiative is an educational campaign to raise the level of awareness of Axis customers on VoIP, and the establishment of online and hotline support.[4]

As of the submission of this report, the proposal submitted to the ministry is still pending as the ministry awaits its annual fiscal budget to be endorsed by the Indonesian Parliament. Should this result in partnership, the income to be generated by the IP PBX installation and after sales services will help sustain the post-project activities and provide financial support to ITMN, which in turn will also support ODC.

However, the proposal submitted to Axis seems to have been rejected.

[3] Please look at Annex 4

[4] Please look at Annex 3

In the future, ODC and ITMN hopes to bring more users in by showing the results of an in-depth evaluation, on cost savings and activities enhanced by IP PBX implementation. ODC will develop a cost-benefit analysis tool to measure how much the same calls would cost if they are dialed using VoIP instead of PSTN or cellular, and the resulting potential cost saving against the equipment purchased, operation and maintenance costs, and the bandwidth used by the system. The tool will be based on any known available standard adopted or built by telecommunication business engagements providing VoIP service.

Annexes

Annex 1: Quality of Service of Codecs suitable to VoIP model applicable to Developing Countries

Annex 2: Komputer Bekas pun Jadi (Even used computer will suffice)
http://majalah.tempointeraktif.com/id/arsip/2009/02/02/TI/mbm.20090202.TI129386.id.html

Annex 3: ODC's solution for Axis

Annex 4: Proposal Pengembangan VoIP Rakyat bersama RISTEK (Proposal to develop VoIP Rakyat with Indonesian Ministry of Research and Technology)

Annex 5: VQ manager Product Help

Annex 6: Op-ed: How to make Telkom remain competive.
http://www.thejakartapost.com/news/2009/06/29/how-make-telkom-remain-competitive.html

Annex 7: Financial Report

Annex 8: Manuscript of VoIP Book

Annex 9: 13 March 2010 – VoIP Workshop at STIKI Malang, Jawa Timur, Indonesia.

Annex 10: 17 March 2010 – VoIP Workshop at SMK YASTI, Cisaat, Sukabumi, Jawa Barat.