A Voice Over IP Deployment Solution for Public Social Institutions of Burkina Faso

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Abstract. Classically, social institutions are provided with a PABX-based unified and shared telephony service. With the mobile communications development, the organization of the telephony service is disrupted in favor of individual use. Mobile terminals capacities and usage development increase this trend. These terminals are more convenient in a social context which requires the ability to communicate anytime and anywhere. One non negligible perverse effect of this individual use of telephony service is the use of workers' private resources for their work. To face this practice, we analyzed Voice over Internet Protocol opportunity for low-budget institutions, designed a low-cost solution which offers a unified and shared access to GSM networks and made propositions about quality of service and security challenges.

Keywords: Voice over IP · SIP · Elastix · Softphone · IP phone · GSM

1 Introduction

The use of individual mobile terminal for accessing voice or data services has considerably increased in many countries of Sub-Saharan Africa [1]. In lot of social sectors of activities such as universities and hospitals, the workers often resort to their private individual mobile terminals for the public service. This is mainly the fact of the unsuitability of the classical PABX-based telephony. The extension and the maintenance of the classical PABX-based telephony infrastructure are expensive for social institutions. Though the classical PABX infrastructure could be replaced by an IPBX infrastructure, the available commercial products are often expensive and covered by licenses. Also, these telephony infrastructures do not natively well accommodate with the mobility requirement of the workers. However, leaning on computer networks which construction will remain a requirement for institutions, it is possible to build a voice service solution which meets the realities of social institutions. Designing a Voice over Internet Protocol (VoIP) deployment solution, of a low-cost institutional access to telephone service which could bring mobility is the main objective of this paper. A VoIP solution based on open software is built to interconnect a diversity of digital terminals (IP phones, PCs, smartphones and tablets) with interconnection to GSM networks. The proposed VoIP deployment solution is accompanied by propositions that address security and Quality of Service (QoS) management challenges.

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2 Overview of VoIP Technology

VoIP [2] is based on the principle that the voice can be digitized and processed as any digital data. Unlike traditional circuit switching telephony, VoIP uses packet switching and IP protocol. In IP telephony, instead of a connection, a session is established between the source and the destination so that there is no dedicated channel. The digitized voice samples, in the form of IP packets, transit over the network by following different paths to the destination where they are processed to deliver an analog voice signal to the user.

Several protocols manage the communication in VoIP. There are usually grouped into communication management protocols and voice transport protocols. Regarding the communication management, ITU-T developed the recommendation H.323 [3]. H.323 takes into account signaling, codecs negotiation and information transmission. As for the signaling management, Internet Engineering Task Force worked on the standardization of Session Initiation Protocol (SIP) [4]. SIP has in charge the management of sessions establishment, release and modification. Each participant of a session has a unique identifier called SIP Uniform Resource Identifier (URI). The required entities for SIP protocol are: (i) a proxy server that receives and processes client requests and routes them to other servers; (ii) a redirect server that handles the translation of SIP addresses; (iii) a user agent that is a software on a user equipment or within an IP phone capable of transmitting and receiving SIP requests; (iv) and a registrar that maintains an SIP URI location database where each user is registered. Other protocols are grafted to SIP to provide additional services; e.g., the Media Gateway Control Protocol [5] used for the interconnection between IP networks and traditional telephony systems. For voice data transport, commonly used protocols are Real-Time Transport Protocol (RTP) [6], Secure RTP (SRTP) [7] developed to address security requirements, Real-Time Transport Control Protocol (RTCP) [8] for the control of the RTP stream and its secure version, Secure RTCP (SRTCP) [7].

The integration of voice data in a computer network is changing exponentially network load [9]. Compression algorithms are therefore essential to deal with the possible lack of resources to carry the large volume of data. The most used ITU compression algorithms (i.e.; codecs) for VoIP are G.711 (PCM), G.729, G.723.1 (MP-MLQ) and G.723.1 (ACELP) with respectively expected Mean Opinion Score (MOS) of 4.1, 3.92, 3.9 and 3.65 [10]. MOS is a metric that depends on the codecs and is useful for the measurement of the quality of VoIP connections. The choice of a codec is a trade-off to make between a desired QoS and available bandwidth.

3 VoIP Service Usages in Sub-saharan Africa

VoIP technology offers several types of services including the traditional and most used service of calls transmission and reception. If a favorable environment of electronic communications development was guaranteed, VoIP would contribute much in the reduction of the call service cost for the end-user. We would observe the development of local operators with the building of their private IP networks for commercial purpose. Unfortunately, VoIP technology, as an opportunity to reach at low cost the universal service for all, has suffered from insufficient and inadequate politics in Sub-Saharan Africa. Despite its advantages by providing telephony access at lower cost to providers and consumers, VoIP was prohibited in many African countries [11]. Until 2005 in South Africa, VoIP was reserved for Telkom, the second national operator and the under-serviced area licensees [12]. Fortunately, since February 2005, the lifting of the restrictions on VoIP opens up a vast number of opportunities and will reduce the costs of voice transmission for both network service providers and consumers [13]. In Burkina Faso, the business opportunity of VoIP technology is still not developed even if the legal environment is favorable [14]. The consumers generally use mobile operators' offers for their telephony needs. Only a privileged few resort to VoIP for international calls using software like Skype. As high rate Internet service is not always satisfactorily offered, this latter usage of VoIP call service remains very marginal and is locally unstructured. A more structured usage of VoIP is that oriented at destination of a group of persons linked by their professional or social activities. VoIP technology has been investigated as a solution to bridge the digital divide to service a specific need of a particular rural community in South Africa [12]. In Burkina Faso, the use of VoIP telephony service in an enterprise or public environment already occurred but is, as the classical traditional PABX, limited to utilization inside offices. In this paper, we propose an open and largely accessible VoIP deployment solution for a usage in a context of high populated social institutions such as universities and hospitals.

4 Proposed VoIP Deployment Solution

The proposed VoIP deployment solution is designed to connect both traditional VoIP terminals (PC, IP phone) and emerging mobile terminals (smartphone, tablet). Thus, users would be able to use their favorite terminal to access services. It also integrates an interconnection with any GSM network. So, the workers would benefit from a unified access to calls outside their network at the expense of their institution. It is designed with open technologies so that its cost is accessible to low budgets.

4.1 Technical Architecture, Services and Cost

A simplified technical architecture of the proposed VoIP deployment solution is depicted in Fig. 1. The network infrastructure is composed of wired Ethernet local area networks with extensions to WiFi components. At the core of the solution is Elastix [15], the server of VoIP services. Elastix is an open software selected for its multiple advantages: it operates with a diversity of protocols (SIP, H.323, IAX, RTP, RTCP) and codecs (G.711, G.729, G.722, G.723.1 …). If hosted on a server hardware with at least equal performance as HP Proliant DL120, 2.5 GHz Xeon, 8 GB of RAM, Elastix could allow 368 simultaneous communications and up to 3000 SIP accounts. Elastix can exchange services with Personal Computers (PC), IP phones, smartphones and tablets. Client software called softphones must be installed on PCs, smartphones and tablets. In Table 1, selected IP phones and softphones for the solution are given. The architecture integrates a firewall in order to filter accesses to the server. Additional

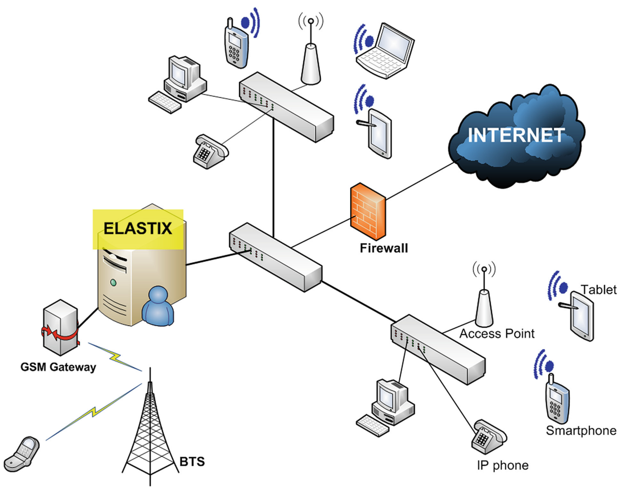

Fig. 1. A simplified technical architecture of the proposed VoIP deployment solution.

Table 1. Client software of the proposed VoIP server

Type of terminal	PC	IP phone	Smartphone/tablet
Softphones/IP phones	X-Lite, Ekiga	Snom 320, Snom 370	Csipsimple

security configurations that should be made are given in Sect. 4.2 together with QoS issue. The VoIP server is interconnected with a GSM network via a gateway like Portech MV-374, a device that can bear four SIM cards. The configuration of the gateway for Elastix is simple and a filtering can be made to authorize selected call numbers if needed.

The proposed solution has been implemented and several services successfully tested: visiophony, call transfer, audio-conference and instantaneous messages. To materialize it, a prior investment in the computer network construction is required. In Table 2, assuming no investment in IP phones and engineering, an estimation of how much could be a minimum deployment of the proposed VoIP solution is given.

4.2 Quality of Service Management and Security Challenges

Network availability and security are major constraints to the integration of VoIP in a computer network. To face these constraints, we recommend some measures for a satisfactory VoIP service. Firstly, we address the problem of the power grid instability. With the traditional circuit-switched telephone network, even when there is power outage, the service remains operational because the power supply is routed by the operator. Such a technique is possible with the IEEE 802.3af standard [16] that provides power to devices over the Ethernet cable. However, as this standard is not open,

Table 2. Cost estimate for a minimum deployment of the proposed VoIP solution

Product	Characteristics	Quantity	Unitary price	Total
Server hardware	HP Proliant DL 120 G5, RAM 8 GB, Hard disk 800 GB	1	660	660
GSM gateway	Portech MV-374, 4 SIM cards	1	1270	1270
Server software	Elastix	-	0	0
Softphones	Ekiga, X-lite, Csipsimple	-	0	0
Overall amount	-			1930 $US

its deployment may be of a prohibitive cost for the targeted institutions. For this reason, we propose the deployment of a solar kit to power the server room.

Secondly, we address the problem of the network bandwidth availability. VoIP is very sensitive to congestion. Low bandwidth is a source of network congestion and causes delays and packet losses. Delay is a critical parameter in VoIP. The phase jitter is caused by temporary network congestion and changes in packets routes. It can be damaging to QoS. To address this bandwidth resource constraint, we propose QoS management solutions, emergency plans and disaster recovery. The establishment of a QoS management policy includes the following three stages: characterization of traffic, classification of flows and definition of stream processing rules favoring voice stream. This last step will be to determine the routing priorities or minimum bandwidth levels for certain applications. With Diffserv (Differentiated Services) [17] architecture, every class of service is subjected to a special treatment. The emergency plan and disaster recovery aim at ensuring the flow of communication and especially the availability of phone equipment in case of network congestion or electrical failure.

Thirdly, VoIP is particularly susceptible to Deny of Service (DoS) attacks. As part of the proposed solution, the firewall will minimize the traffic entering the server and may limit DoS attacks. Also, we recommend the deployment of anti-hackers modules to protect Elastix and prevent hackers from attacking the main services.

Finally, to overcome communications confidentiality, we recommend encryption-based protocols such as SIPS [18] and SRTP respectively for signaling flow and voice flow encryption. SIPS also prevents from risks of identity theft.

5 Conclusion

This paper dealt with a low-cost VoIP deployment solution for social institutions such as universities and hospitals. The solution allows the users to use their smartphones or tablets to access VoIP services. An interconnection with GSM networks is made to offer a unified access to call services at the expense of the institution. The proposed VoIP solution could contribute significantly in the fight against poverty in institutions where the workers are constrained to use their own resources for the public service.

As perspectives, this study will be refined especially by sizing needs and capabilities of equipments. The impact of the integration of voice data in the computer network will be studied and possible extension of the proposed solution will be made.

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References

1. Guy, Z., Christine, Z.-W.Q., Jenny, C.A., François-Xavier, R., Annie, C.-L., Samir, S., Tangui, B., Véronique, P., Guillaume, B.: La téléphonie mobile dans les pays en développement: quels impacts économiques et sociaux? Secteur Privé & Développement, Proparco, no. 4 (2009)
2. Black, U.: Voice Over IP. Prentice-Hall, Upper Saddle River (2000)
3. Liu, H., Mouchtaris, P.: Voice over IP signaling: H.323 and beyond. IEEE Commun. Mag. **38**(10), 142–148 (2000)
4. Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., Schooler, E.: SIP: Session Initiation Protocol, RFC no 3261 (2002)
5. Anquetil, L.-P., Bouwen, J., Conte, A., Van Doorselaer, B.: Media gateway control protocol and voice over IP gateways. Alcatel Telecommun. Rev. **2**, 151–157 (1999)
6. Schulzrinne, H., Casner, S., Frederick, R., Jacobson, V.: RTP: a transport protocol for real-time applications, RFC no 3550 (2003)
7. Baugher, M., McGrew, D., Naslund, M., Carrara, E., Norrman, K.: The secure real-time transport protocol (SRTP), RFC 3711 (2004)
8. Ott, J., Wenger, S., Sato, N., Burmeister, C., Rey, J.: Extended RTP profile for real-time transport control protocol (RTCP)-based feedback (RTP/AVPF), RFC no 4585 (2006)
9. Nutt, G.J., Bayer, D.L.: Performance of CSMA/CD networks under combined voice and data loads. IEEE Trans. Commun. **30**(1), 6–11 (1982)
10. Mehta, P., Udani, S.: Voice over IP. IEEE Potentials **20**(4), 36–40 (2001)
11. Cohen, T., Southwood, R.: An Overview of VoIP Regulation in Africa: Policy Responses and Proposals, Commonwealth Telecommunications Organization (2004)
12. Chetty, M., Tucker, B., Blake, E.: Using voice over IP to bridge the digital divide - a critical action research approach. In: Southern African Telecommunications Networks & Applications Conference, South Africa (2003)
13. Chetty, M., Blake, E., McPhie, E.: VoIP deregulation in South Africa: implications for underserviced areas. Telecom Policy **30**(5–6), 332–344 (2006). Elsevier Ltd.
14. Autorité de Régulation des Communications électroniques, Déclaration de politique sectorielle des télécommunications de juillet (1999). http://www.arce.bf
15. Elastix official web site. http://www.elastix.org
16. Schindler, F.R.: U.S. Patent No. 7,930,568. Washington, DC: U.S. Patent and Trademark Office (2001)
17. Babiarz, J., Chan, K., Baker, F.: Configuration guidelines for DiffServ service classes, RFC 4594 (2006)
18. Salsano, S., Veltri, L., Papalilo, D.: SIP security issues: the SIP authentication procedure and its processing load. IEEE Network **16**(6), 38–44 (2002)