

Challenges of Integrating a VoIP Communication System on a VSAT Network

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Abstract — This paper summary whole research aimed to study integration of a VoIP communication system on the VSAT network of ASECNA, which consisted to propose a solution based on VoIP technology, allowing ASECNA's workers to communicate, from different countries and working places. However, since the ASECNA's network is based on satellite transmission, obvious difficulties (such as latency, packet loss, etc.) appear in implementing a VoIP system over this network, because VoIP is demanding in terms of quality of service (QoS).

Thus, in this study, through simulations and tests in real situations, we determined technical choices to be made (software, codecs, and protocols) and corrective actions to take for the successful implementation of a VoIP based system in ASECNA's context.

Keywords – ASECNA, Asterisk, IAX, VSAT/satellite, QoS, VoIP;

I. INTRODUCTION

With technological developments and the current trend of network convergence, more and more companies make the choice of VoIP technology to optimize the use of their network resources and reduce communication costs, while benefiting from diversification of means of communication (telephone, video conferencing, instant messaging, etc.)

Integrating a VoIP communication system in the VSAT network ASECNA (Agency for the Safety of Air Navigation in Africa and Madagascar) is a hot topic for this great company, where VoIP technology ("Voice over IP" or "VoIP") is operated at a low level; then as implemented on a wider scale, it could improve the internal communication channels of the company, including between agents or officials of ASECNA working in different countries.

It will therefore study the possibility of setting up a solution based on a PBX (Private IP Branch eXchange) or IP PBX to take advantage of the many opportunities offered by VoIP technology, and opportunity to take advantage of the current capacity of the telecommunication network of ASECNA. To do this, we will begin by presenting the context and the motivations of our project, the problematic, the intended objectives and that

our methodology of work. We will then identify and analyze needs of integrating VoIP system on VSAT network. In order to conclude on the feasibility of such a project, we will proceed to the implementation of VoIP system, and finally analyze and interpret the results obtained with the study of a few cases.

A. Context and motivations

1) Issues of quality of service of a 3G network

As part of communications between the centres of ASECNA, telephone calls are made at two levels:

- operational communications, as part of the supply of the services of the air traffic, called ATS / DS (Air Traffic Service / Direct Address), and the telephone of service, for exchanges between units of service of the different countries;
- not operational communications, which allow administrative structures of the different countries (general administration, human resources, finances, paierie, etc.) to communicate between them or with the Seat, as part of their activities. For ATS / DS, ASECNA has point-à-point (devoted lines) telephone network. But this network is strictly used for the services of the air traffic, and cannot be used for not operational communications. For the telephone of service, every centre has only the single line allowing him to call units of service of another centre, and therefore the single communication is possible at the same time. However, this line is not either recommended for administrative communications.

So, for not operational communications between sites, fixed lines rented to the historical operators, and, more often, subscriptions taken to the operators of mobile telephony are used; what procreates a cost mattering for the Agency.

2) Constraints of VoIP systems

Unlike the classical telephony, which works in mode circuits, the VoIP operates in mode packets, which induces several constraints to offer an experience of satisfactory communication. main constraints having most impact on systems VOIP are based on :

- quality of codec

The scanning process and compression of voice, in VoIP networks, is carried out by codecs (encoders/decoders)

codec	Standard	Débit (kbit/s)	Fréquence (kHz)	Windows size (ms)	MOS (Mean of Opinion Score)
G.711	UIT-T	64	8	20	4,1
G.723.1	UIT-T	6,4	8	30	3,9
G.728	UIT-T	16	8	2,5	3,61
G.729	UIT-T	8	8	10	3,92
iLBC 20	—	15,2	8	20	4,14
iLBC 30	—	13,33	8	30	—

- The Latency

The values recommended by the ITU-T Recommendation G.114 are:

- less than 150 ms: good quality;
- between 150 and 400 ms: acceptable;
- beyond 400 ms: unacceptable, except in exceptional cases

- The jitter (RFC 3393)
- The load of network
- The echo

A VoIP communication involves two types of protocols: signaling protocols and the transport protocols :

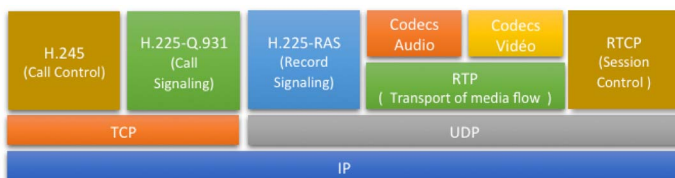


Figure 1. Stack of the protocol H.323

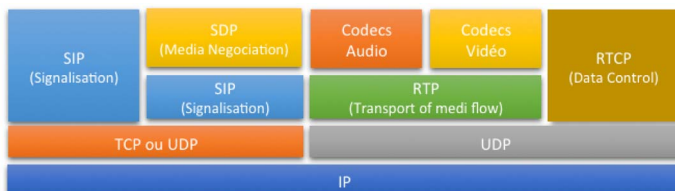


Figure 2. Stack of the protocol SIP

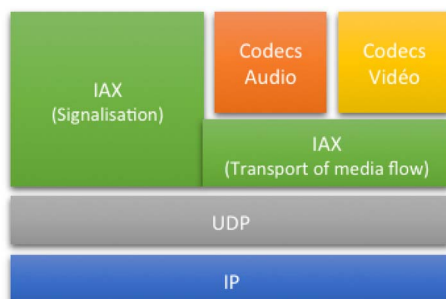


Figure 3. Stack of the protocol IAX

The VoIP presents many advantages, such as the reduction of the costs or the optimization of resources, thanks in particular to the convergence telephone network - computer network. In addition, the use of open standards, such as the SIP protocol, allows interoperability and facilitates the production of a large number of solutions, several of which are also free of charge. It has some drawbacks as the quality of the voice, which is lower than that of the classical telephony, or still further the

problems of security, inherited from the weaknesses and vulnerabilities of the IP protocol and of computer systems

B. Methodological approach

To achieve the study of the establishment of a communication system VoIP, it is necessary to take account of the existing network infrastructure. As well, we will begin this work by the study of the telecommunications network of the ASECNA and the study of its current communications systems, which we will highlight the constraints and limits.

Then, on the basis of the elements obtained in the previous phase, we will propose solutions Voice on IP among which we will retain the more suitable, depending on a number of criteria. Then, we will propose an architecture for the implementation of this solution to the scale of any the ASECNA.

At the same time, we will discuss the codec that will allow to obtain an optimal quality of listening, while optimizing the use of the bandwidth. To do this, but also in order to verify the effectiveness of the system, we will achieve tests in "Laboratory", from virtual machines, and tests in a real situation, performing phone calls with people located on different sites of the ASECNA. These tests will assess various parameters of quality of service (QoS - Quality of Service) and to take the right decisions. Taking into account the vulnerabilities inherent to computer systems, on which are based VoIP systems, we propose some basic security measures at the end of this work, we want to provide a study sufficiently developed, which will serve as the basis for the deployment of a system of VoIP communication across the ASECNA.

As well, we will have to present clear and concrete results on the feasibility of the project, the efficiency of the system, the use of network resources, of hardware resources.

II. ANALYSIS OF THE VDI NETWORK AND IDENTIFICATION OF NEEDS

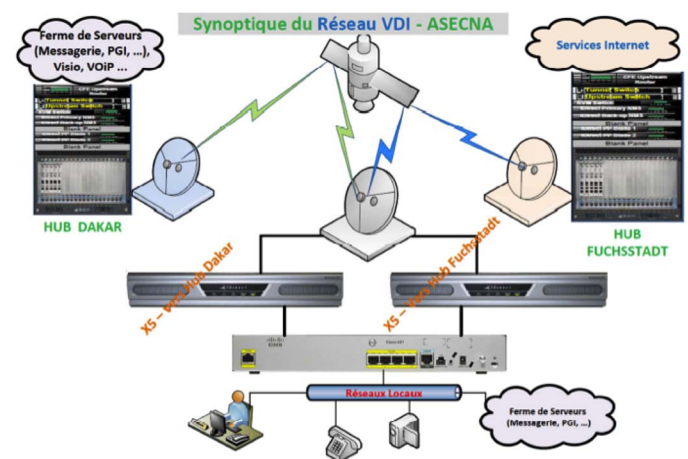


Figure 4. VDI (Video Data Image) network of ASECNA4.

A. Study of the existing network

The VDI network is built on a star topology. It is composed of two (2) Access:

- An intranet access, which supports the services Corporate, and whose hub is in Dakar (Senegal);

- a Internet access, which supports the Internet services, and whose hub is to Fusschtadt (Germany).

As shown in figure4, the system, on each site, is composed mainly of the following equipment:

- a Cisco router 880, which provides the interface between the local network of the Center and the two (2) access the Intranet and the Internet;
- the iDirect system, which includes the modem-router iDirect X5 and other equipment not represented on the synoptic (upstream switch, switch Tunnel, Protocol processor, etc.), which prepare the received packets to send to the radio part;
- the antenna: that of the hub is an antenna of a qualified station B (11 m in diameter); those of the sites are the antennas on qualified stations F2 (7.3 m in diameter) or F1 (3.7 or 3.8 m in diameter);
- the satellite: the ASECNA leases of frequency bands on the Intelsat satellite 10-02@359° E, on the transponders 20/20 and 23/23.

The telephone networks currently existing at the ASECNA have several limitations.

First of all, the ATS/DS (Air Traffic Services / Direct Speech) is a dedicated service to air traffic, and therefore cannot be used in the framework of Communications not operational, as is our objective. In addition, it does not allows you to join that adjacent centers, while we want a system which covers the whole extent of the Agency. Finally, it is also open to the centers adjacent non-ASECNA, while our system wants to be a solution of internal communication to the Agency.

The telephone of maintenance presents virtually the same limitations as the ATS/DS.

On the technical level, at the level of the Memotec (multiplexers which support the ATS/DS and the phone of the maintenance), the creation of a subscriber involves the addition of a hardware (map), which limits the flexibility and the number of subscribers to the number of slots (location of cards) available.

Concerning the telephone networks local, given the improvement of the telecommunications network of the ASECNA, through the establishment of the VDI network, it would be possible to reset on table the abandoned project for the interconnection of switches. But the cost of the licenses ALCATEL being very high, such a project for the 30 sites ASECNA concerned would be very expensive.

With respect to the videoconferencing system, in spite of its significant performance, it does not offer a very high flexibility, in terms of Clients software (Softphones) and equipment (IP Phones) supported. In addition, it is a system that is particularly optimized for video conferencing, while we are looking for a solution for a telephone service. Finally, it requires a specific equipment, which makes it expensive.

As well, we should therefore be a more flexible system, but also more economical.

B. Modelisation of the solution

In examining this comparative table below, it becomes visible that in spite of the fact that Asterisk has no native GUI, it presents many functional benefits, the possibility to modify the source code for the adapt to our needs (free software) and especially the free. In addition, it is easy to find on Internet discussion forums concerning Asterisk, in case of a problem

	Skype	Asterisk	3CX Phone System
Platform	Internet	Linux	Windows
In a LAN	no	yes	yes
Sécurité	yes	yes	yes
Extensibility	No	yes	yes
Protocols	private	SIP, IAX2, H.323, SCCP, MGCP	SIP
Audio codec	G.729, SVOPC, SILK, Opus	G.711, G.722, G.723.1, G.726, G.729, GSM, iLBC, Speex, etc.	G.711, G.722, GSM, Speex, iLBC
Video Codecs	VP7, H.264	H.261, H.263, H.263+, H.264, VP8	H.264, VP8
terminals	Skype	All type	3CX Phone (logiciel), Fanvil, HTEK, snom, Yealink, Aastra, Cisco, Gigaset, Panasonic, Polycom (material)
Graphical Interface	yes	No	yes
Connectivity network (fixes/mobiles)	yes	yes	Yes
License	Not free	Free	Not free
Cost	free	free	Not free

Table 1: Comparison of Three VoIP solutions

Given the star topology of the VDI network of the ASECNA, two (2) opportunities we have to:

- a centralized architecture, with a single server to Dakar (the hub);

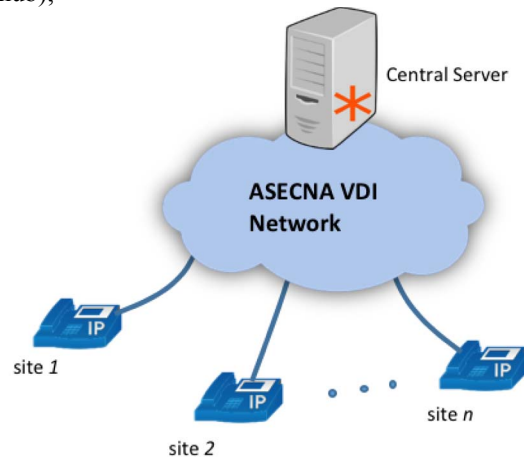


Figure 5. Centralized architecture

- since the server is central, even the local communications, between two (2) clients of a same site, must go back up to Dakar for be treated; which leads to an increase in the latency and the unnecessary consumption of bandwidth;
- the large number of configurations to achieve for all subscribers of all sites;
- the solicitation important from the server to be able to deal with all the communications (local and inter-sites) of the 34 sites;
- the unavailability of this single server (and its eventual backup) penalizes the entire system, even the local communications;

- a distributed architecture, with a server on each site.

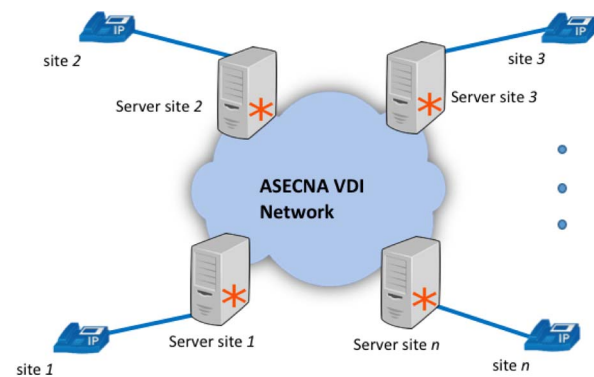


Figure 6. Distributed architecture

- the bandwidth of the network VDI is consumed for the inter-site communication;
- the configuration of the server is reduced, since it only concerns the subscribers of the site and the connections with the other sites;
- the server load is reduced, since it only manages the communications specific to its own site;
- the unavailability of a server only affects the site concerned;

C. Simulation environment

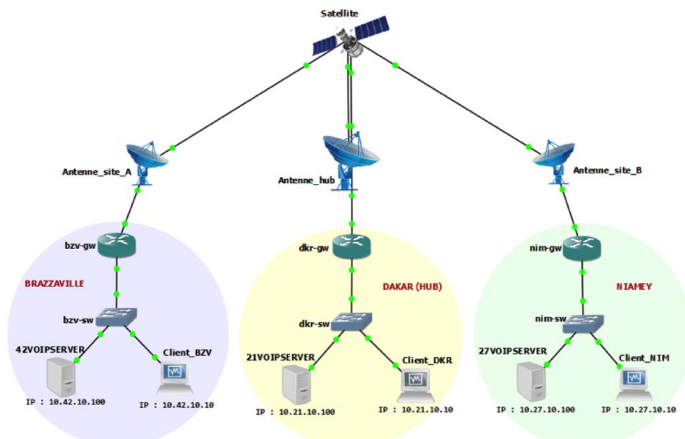


Figure 7. Architecture of simulation

On each site, we have:

- a server machine (42 VOIPSERVER, 21 VOIPSERVER, 27 VOIPSERVER) on which are installed the operating system Ubuntu Linux 14.04, the Asterisk IPBX 13.9.1 and the analyzer of Wireshark Packet;
- a client machine (client_BZV, CLIENT_DKR, CLIENT_nim) on which are installed the Windows 7 operating system and a generator of SIP Call (SIP StarTrinity test);
- a switch and a router that provides the interface between the local network and the radio part (modem antenna) of each site;
- an antenna, representing the radio part (modem antenna) of each site. This antenna is in reality simulated by a virtual router on each site, and represents the Modem Router iDirect X5. The three (3) routers use the dynamic routing protocol RIPv2, as is the case in the VDI network between the modems of the sites and the hub.

Finally, the satellite allows to link the three (3) sites. It is simulated by a virtual computer hosting the application WANem, which allows you to emulate a WAN network and to

reproduce the conditions of transmission by satellite. Each link between the antenna and the satellite represents a satellite connection.

III. ANALYSIS AND COLLECTION OF RELIABLE PARAMETERS FOR THE DEPLOYMENT OF VOIP SYSTEM

A. Evaluation of the latency

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Statistiques Ping pour 10.200.0.2:
Paquets : envoyés = 100, reçus = 100, perdus = 0 (perte 0%),
Durée approximative des boucles en millisecondes :
Minimum = 507ms, Maximum = 819ms, Moyenne = 613ms
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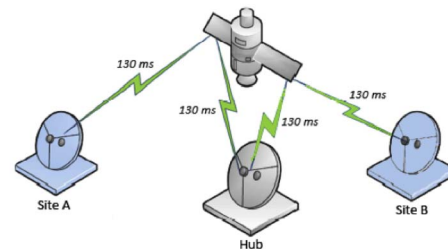


Figure 8. latency due to the satellite

We can therefore anticipate a latency of approximately 300 ms for a communication between a site and the hub, what is tolerable, according to Recommendation ITU-T G.114. By contrast, the latency of site to site, which could exceed the 600 ms, could be detrimental; but the tests in real situation will allow us to better we speak on this issue

B. Determination of the codec to use

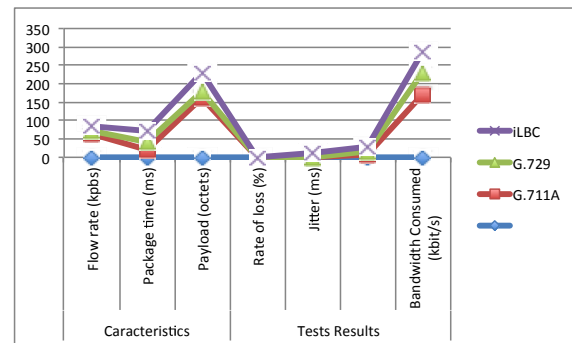


Figure 9: Comparison of Three VoIP solutions

These results show that the iLBC codec is the more economical in terms of consumption of bandwidth; but its jitter higher leads us to him prefer the G.729 codec, which is also economic in bandwidth and presents the jitter is the lowest. That is why we opted for the latter for the implementation of our solution.

C. Determination of the signaling protocol to use

We generated calls G.729 Concurrent, in growing numbers, from site A to Site B on the one hand, the SIP Application Test is used to generate the calls; on the other side, it serves to respond. The communication is programed to stop automatically after 60 seconds.

In a first time, we establish a link inter-server with SIP, and then we will resume with a link using IAX in the trunk mode. Finally, we evaluate each time QoS and the bandwidth used.

The poor quality of the SIP communications from 3 simultaneous calls is probably due to the fact that the network (in particular the routers) is not optimized for the voice. However, despite this insufficiency, we have been able to perform 10 communications simultaneous IAX, which is more, consume less bandwidth than RTP in the same conditions. This can be explained by the fact that with the multiplexing of communications carried out by IAX, the number of packets and the volume of data traveling on the network are much less important.

For these reasons, we opt therefore for the Protocol IAX for signaling inter-servers.

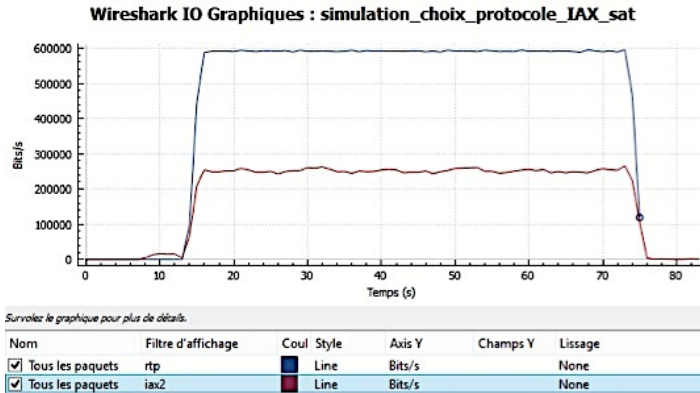


Figure 10. Bandwidth of RTP and IAX, used for 10 simultaneous communications

D. Determination of the Bandwidth

The 34 sites share 9 carriers of 2 Mbit/s in uplink, or 4 sites by carrier. In reality, it is the flow of information, or Information Spleen, obtained with a modulation 8-QAM and a FEC of 4/5.

The iDirect technology, thanks to its improved version of the standard DVB-S2, allows to reach approximately 95% of the flow of information for IP traffic [W23], either:

$$2 \text{ Mbps} \times 0.95 = 1.9 \text{ Mbit/s.}$$

We ask then as assumption that at the time charged, the voice traffic must not exceed 50% of the bandwidth IP, either:

$$1.9 \times 0.5 = 0.95 \text{ Mbit/s} = 950 \text{ kbit/s.}$$

This is tantamount to saying that each site must not consume, for voice traffic, more than:

$$(950 \text{ kbit/s}) / 4 = 237.5 \text{ kbit/s}$$

Now we are to determine, the number of simultaneous calls that it is possible to perform from each site to other sites, in function of the limit of 237.5 Kbit/s.

E. Calculation of the number of simultaneous calls possible

$$n = \frac{20 \cdot 10^{-3} \times 237.5 \cdot 10^3}{8 \times (42 + 20)} \approx 9 \text{ Simultaneous calls}$$

In summary, therefore, it is possible to perform between 9 and 21 simultaneous calls, depending on if the calls are all going to different destinations or, on the contrary, to the same destination.

By combining the architecture, the protocols and the codec selected, with the bandwidth values calculated previously, we get the following block diagram the VoIP system:

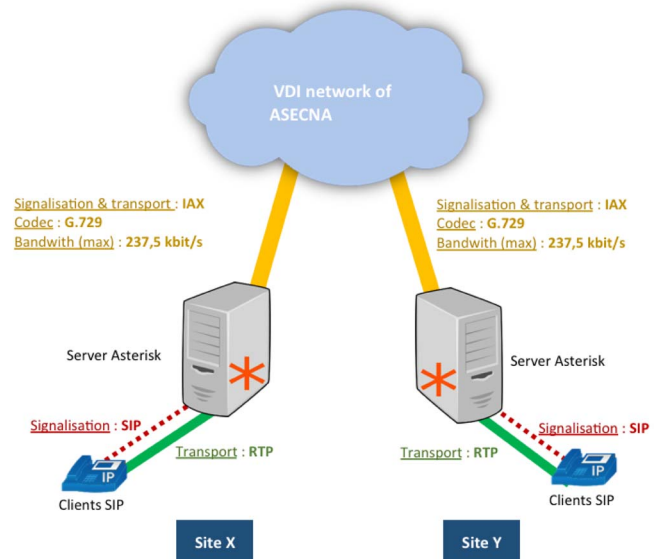


Figure 11. Final network with system VoIP communication

IV. DISCUSSION AND SOME CHALLENGES OF THE APPLICATION AND THE PERFORMANCE ANALYSIS

Case study : Call between Dakar and Brazzaville and evaluation of the quality of service :

to assess the quality of service of the connection, we noted the call statistics with the Protocol IAX every 20 seconds, during the conduct of the appeal by running the following command from the Shell Linux:

watch -n 20 'asterisk -Rx "iax2 show netstats" '

referring to this formula, where the number n_{pp} of lost packets is equal to $n_{pp} = n_{Lost} + n_{Drop} + n_{000}$ and to assess the rate of loss of packets τ_{pp} , we apply the formula $\tau_{pp} = \frac{n_{pp}}{n_{pp} + n_{reçus}} \times 100$ and obtained the following results:

- Latency

Time (sec)	20	40	60	240	260	280	Min.	Max.	Moy.
RTT (ms)	531	519	521	566	577	547	519	765	572
Latence (ms)	266	260	261	283	289	274	260	383	286

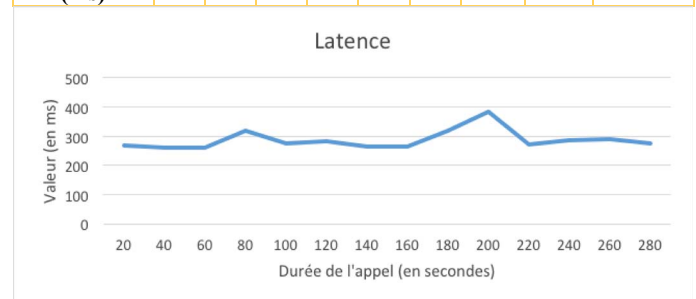


Figure 11. Latency on the call Dakar – Brazzaville

Our surveys we show a higher latency to 200 ms, but less than 400 ms, with an average of 286 ms. This value is acceptable.

- Jitter

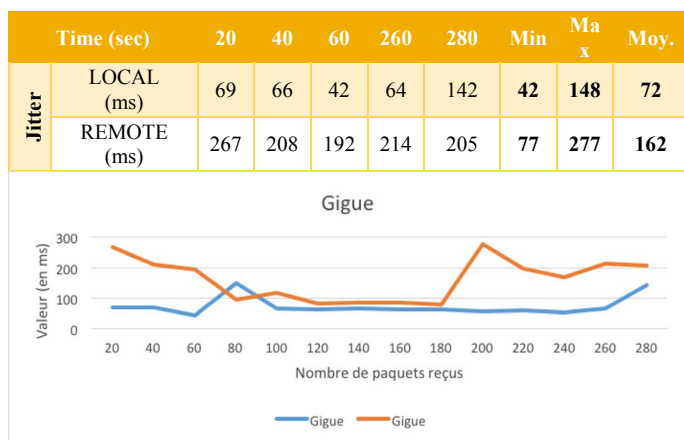


Figure 12. Jitter on the call Dakar - Brazzaville

The side "local", the average jitter is 72 ms. On the other hand, the "remote", the jitter is quite variable and on average above the 100 ms (162 ms), with strong fluctuations after 3 minutes of conversation. This fluctuation may be due to an increase of the load on the network.

- Rates of packet loss

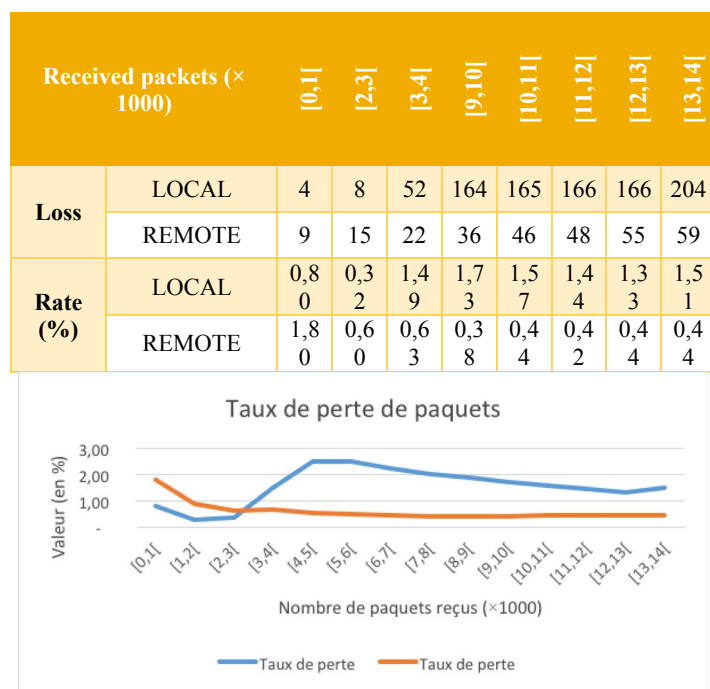


Figure 13. Loss rate on the call Dakar - Brazzaville

The rate of loss, the remote side is almost negligible, with an average of 0.44%. By contrast, the local side, it is medium with a value which exceeds 1 per cent (1.51%). This is characteristic of a network medium loaded.

As well, we make the following recommendations, in order to obtain a better quality of service for a possible large-scale deployment of this VoIP system:

- Put in place a policy QoS on the WAN to ensure dynamically from the bandwidth for VoIP traffic. It will, however, take care to define a MIR (Maximum Information Rate), maximum limit

of bandwidth, that the whole of the VoIP communications of the network should not exceed;

- put in place a QoS policy in local networks, to prioritize the VoIP traffic on the routers providing the interface between the local network and the network VDI, and on the switches used in local networks, if this feature is available;
- Create VoIP VLAN in local networks, i.e. sub-virtual networks intended only to VoIP traffic, to avoid congestion;
- Modernize the local networks of sites which could potentially have a hardware (switches, cables) aging or obsolete;
- put in place a Rule Security on the firewall to prevent that the incoming and outgoing traffic by the UDP port 4569, which is the port used by the Protocol IAX, might be blocked; but also to ensure that this traffic originates from a host of confidence.

In addition, concerning the deployment of the communication system studied in this memory, we recommend a phased deployment in three (3) phases:

- Phase 1: Deployment in the principal centers of each fir of the ASECNA (Antananarivo, Brazzaville, Dakar, N'Djamena and Niamey), at Headquarters and at the EAMAC. This will allow to assess the behavior of users, see the impact of this new service on the VDI network, doing analyzes, and then apply the changes/restrictions necessary;
- Phase 2: Deployment in the main centers of the other member countries of the ASECNA, the other schools (ERNAM, ERSI), and the Delegations;
- Phase 3: Deployment in the secondary centers

V. CONCLUSION

We have been able to put in place a system of VoIP communication between the sites of the ASECNA. This result has been made possible by, among others, thanks to the performance of iDirect technologies, on which rests the VDI network of the ASECNA, and thanks to the effectiveness of Free Software Asterisk and its Protocol IAX.

On the basis of the theoretical results obtained, we have implemented the VoIP system on three (3) Sites: The hub (Dakar) and two (2) Pilot Sites (Brazzaville and Niamey). Then we have conducted tests in real situation, performing calls between these sites. In particular, the tests between the pilot sites have found a latency Very Important (between 620 and 750 ms), a jitter high (around 200 ms), but offset by a memory-buffer of compensation of jitter (jitter buffer), and a rate of loss medium (between 0.5 and 1.5%), using the G.729 codec. However, these values have not had a disastrous impact on the quality of communications, since the interlocutors have been able to communicate without providing of efforts.

However, in relation to certain of the high values obtained, particularly concerning the latency, jitter and packet loss rate, we have made proposals for the optimization of the VDI network and local networks of sites ASECNA, mainly through the implementation of QoS policies favoring the VoIP traffic.

Finally, the study could examine how to interconnect the VoIP system with the mobile network, to take advantage of the GFU networks (closed group of users), and have the possibility to reach an agent directly on his mobile, from the VoIP system.

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