Development of Multisite Communication Architecture using VOIP Conferences

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ABSTRACT

To guard and manage maritime territories, a nation must have several stations located at strategic position. This mean, usually, a remote island or location close to border. These station must equipped with radar, AIS receiver and HF/VHF communication devices to interrogate every ships who passed the nation's border. Every stations should be connected to each other by telecommunication media means. The interconnections between stations are using old technologies of telephony. Cost of operation is increased every year.

These paper propose a strategy to migrate from dedicated line of telephony, to VOIP. Communication with VoIP can encourage us to change the old network of point to point dedicated line to more cheap and efficient network, such as MPLS. VoIP can be extended to serve conference. So communication is not just for 2 participants, but able to serve 3 or more participants.

We developed VoIP system that can accommodate one or more participants to join conversation existed, thus forming a conference. We use conference bridge which is shipped together with Asterisk application, to multiplex all voice signal. In order to keep load of server Asterisk low, we use distributed server model.

KEY WORDS: *VOIP, audio conference, Asterisk.*

1.0 INTRODUCTION

A country needs to monitor every ship which passes their territory. Sometimes, they must interrogate or communicate with crew of the ship. Communication's signal will travel from ship to control center through remote station (usually lighthouse). Transmission media between ship and remote station is radio. And transmission media from remote station to control center is using dedicated line from telecommunication operator.

In these modern ages, dedicated line isn't efficient. Telecommunication operator refuses to continue serve this service. It will be replaced by packet switching. There are several packet switching technologies, like Frame Relay and ATM. On top of packet switching, we use MPLS (Multi Protocol Label Switching). MPLS can carry varies of internet data, including voice. This paper will propose a strategy to move from dedicated line to internet based communication without breaking compatibility with standard communication between ship and control center.

To take advantage of this new technology, we will also try to add the capability of conference communication between several control centers. The conference will be used in this scenario: if there is a sick crew, and in control center which responsible for that ship haven't a doctor, they can call another control center which have an attending physician, and create a conference for diagnosing.

2.0 VoIP (VOICE OVER IP)

A basic concept of VoIP is packetizing the analog voice for transferring through internet network. TCP/IP Protocol, the foundation of internet, is not designed for voice or real time packet. When a packet lost, a receiver will wait, and request a packet resend after several seconds. In voice conversation, this concept will not work well. If there is a packet lost, it means some words are missing. And if receiver waits for packet resend, it means the communications falters. VoIP try to overcome these problems.

VoIP is using signaling to establish communication. There is several signaling protocol:

- SIP. SIP (Session Initiation Protocol) uses the syntax of another protocol such as TCP / IP, text based. The SIP specification can be read in RFC 2534 [6]. SIP is a protocol layer application that uses port 5060. SIP can be carried on UDP or TCP. SIP is used to establish, modify and terminate multimedia sessions such as Internet phone calls. SIP does not carry media between end devices. SIP use RTP to transmit the media. The conversation between two SIP devices is possible without SIP server.
- H323. ITU develops H323 to provide IP with video conferencing capability over LAN. But, many companies enjoy the ability of voice. H323 is used in the telecommunication operator. Society and typical user use SIP. The port used is at 1720.
- MGCP. MGCP (Media Gateway Control Protocol) has also come from IETF, like SIP. MGCP use port 2427. Standard is on RFC 3435. The aim is to make the simplest job of the post. The post should not make a lot of signaling. Most signaling is done by voice gateway, so it is not possible two SGCP's post are connected directly. We must use MGCP server.
- IAX. IAX (Inter Asterisk Exchange) is developed by Digium for communications among Asterisk's server. The Protocol is open for everyone. IAX uses port 4569 for signaling and media. So it's easier for IAX to traverse firewalls as SIP. IAX trunk can make some session conversations in a stream with a single header. This contributes to a smaller latency, reduces the processing treatment and required bandwidth.
- SCCP. The Skinny Call Control Protocol (SCCP) is the property of Cisco VoIP equipment. It is the default protocol for endpoints on Cisco Call Manager PBX. SCCP use TCP Port 2000. To transfer media, it uses RTP.

We choose SIP because it was developed for the Internet (WAN) not only for LAN. And we do not need video communication. Another reason is there are many choices of software and hardware that provide SIP, so it's easy to find ones with high quality.

3.0 VOICE CONFERENCE

In SIP, a conference is an instance of a multi-party call (RFC4353). In the conference, the post must hear any simultaneous communications. There are some conference models [7]:

- Multicast model. It took advantage of multicast system ability. Each station must have VOIP IP multicast.
- Full mesh. Each item must send the message to all conference participants.
- End system mixing. The client that initiates the conference should mix everything and forward to any participants.
- Centralized server. The server receives the data communications, and distributes them to all client, unless the original sender.

Our basic need is to keep existing equipment; the best solution is Centralized Server. It does not change the IP address, or change the configuration station or router, or changes the client. Just add the server in the network. The processing delay problem can be solved by using powerful server.

4.0 CISCO VOICE ROUTER

For interfacing between legacy communications devices to internet network, we use Cisco Voice Router 2800 series. Cisco routers are equipped with VIC2-E / M module. Actually, the module is to connect between analogue PBX and router. The link is called trunk. There are two trunks: analog and digital. The E & M protocol (Earth & Magneto) is used for signaling between two PBX analog trunks. If there is an analog extension connected to PBX that wants to call extension connected to another PBX remote analog, use the S & M protocol. ISDN signaling is used on digital trunk.

E & M signaling provides state conditions off hook (hook) or on hook (hook). There are 3 types of indication: immediate start, delay start and wink start [8]. There are 6 different types of E & M signaling. Type I to Type V, and SDDC5. SDDC5 is similar with Type V. Cisco does not provide type IV.

5.0 ASTERISK

Asterisk is a software implementation of a telephone private branch exchange (PBX). Like any PBX, it allows attached telephones to make calls to another, and to connect to other telephone services including the public switched telephone networks (PSTN) and Voice over Internet Protocol (VoIP) [11]. Asterisk is free software that is primarily designed to run on Linux. [12]

6.0 DESIGN AND IMPLEMENTATION

After several experiments, we arrive at control center's topology shown in figure 1.

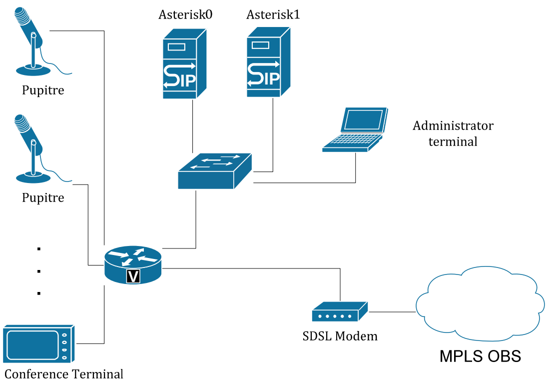

Figure 1: Topology of Control Center

The method depends to Asterisk server. If the Asterisk server fails, the system no longer works. We must add redundant. When there is interruption or failure, Cisco will automatically move on the second server for about 10 seconds. We were using stopwatch to measure the time needed to change from first server to server to second server. We started the stopwatch by removing the UTP cable. I look at the Asterisk console. If all conferences are established, we stopped the stopwatch. The table 1 is the information of time needed to failover with 10 times of tries. There are changes in time because each voice port of Cisco has their own timer to resend their package.

Table 1: Time Required to Change Server

Time (seconds)
10,02
11
9,54
10,87
10,4
10,76
10,93
10,39
10,58
10,94

And for remote station, we are using topology as shown in figure 2.

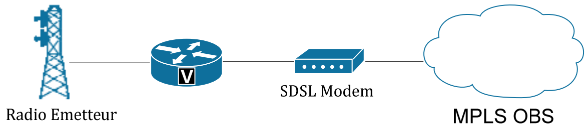

Figure 2: Topology of Remote Station

The Asterisk server is installed in the control center side to ease maintenance and server update.

The steps of Asterisk configurations are:

a. First is /etc/asterisk/sip.conf configuration. The function of sip.conf is to authenticate SIP posts, it is allowed to use the Asterisk resource or not. After changing the configuration, always reload with command sip reload in the Asterisk console.

b. We need the SIP extensions register to Asterisk. To see wether the SIP post is already sign or not, we can check it with command sip show peers in the Asterisk console. If there is an error, then analyze it with either the log or the console message or packets that are exchanged with Wireshark [13].

c. Creates the dialplan extensions. An extension here is not same with definition of traditional PBX ext. Extensions in Asterisk is simply counting and naming action set. The configuration is in the extensions.conf. If we call a number, Asterisk will look for the number on extensions.conf. Because all the communication to the conference room, then everything will move forward on the call conference room.

d. Configuring the conference. The conference use MeetMe application. The configuration is in the meetme.conf. Configure rooms section, and adds the conference room. MeetMe use µLaw for processing. So, if there are some participants uses GSM or Alaw, the processing time will increase to transcode the signal.

Next, is the Cisco's configuration. In the voice interface card (VIC), there is DSP processor for the calculation. If you do not want to share the resource of this DSP, we issued no dspfarm. If we do not use much channel, it can reduce the computational complexity to increase encoding speeds with medium complexity codec.

To mark the package with the IP addresses consistently, we use bind control and bind media. Cisco can change the codec to adapt its connected party. You can configure the priority of the codec with voice class codec. Another class is voice class permanent, it is to put the trunk in permanent connection.

For the called SIP post configuration, we create dial-peer voice. By default, Cisco uses H323, then if we want to use SIP, session protocol SIPv2 command is required. To make conference, Cisco must register Asterisk server. It is the function of sip-ua. It configures the Asterisk server address and username and password.

6.0 ANALYSIS OF COMMUNICATION BETWEEN CISCO AND ASTERISK

In this simulation, we directly connect Cisco and Asterisk using UTP cable. For best results, we do several simulations. We found that the results are consistent, jitter and delay change a bit. The figure 3 below are the results of simulations.

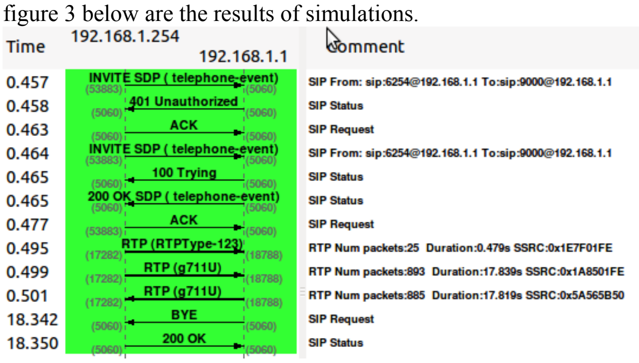

Figure 3: Flowgraph between Cisco and Asterisk

Cisco is left, and Asterisk is at right. To join the conference, the procedure is similar to the procedure name. It started with SIP INVITE from post SIP to the conference room phone number, in this case is 9000. We can protect the room with pin number, but we do not implement it. SIP Status 401 Unauthorized forwarded to the challenge of this pin.

And Cisco forwarded SIP INVITE has been answered with SIP 200 OK Status. In call setup, there is always SDP. This is to negotiate the codec. We use the G711 codec µLaw. After that the communication starts.

Analysing stream from 192.168.1.254 port 17562 to 192.168.1.1 port 22808 SSRC = 0x67301FE

Pack	Sequen	Delta(r	Filtered Jitte	Skew(ms)	IP BW(k	Mark	Status
19	2550	19.99	0.00	0.08	6.40		[Ok]
21	2551	20.00	0.00	0.08	8.00		[Ok]
23	2552	20.00	0.00	0.08	9.60		[Ok]
25	2553	20.02	0.01	0.06	11.20		[Ok]
27	2554	20.00	0.01	0.07	12.80		[Ok]
29	2555	19.98	0.01	0.09	14.40		[Ok]
31	2556	20.01	0.01	0.08	16.00		[Ok]

Max delta = 20.22 ms at packet no. 1737
Max jitter = 0.04 ms. Mean jitter = 0.01 ms.
Max skew = 1.22 ms.
Total RTP packets = 1228 (expected 1228) Lost RTP packets = 0 (0.00%) Sequence errors = 0
Duration 24.54 s (-136 ms clock drift, corresponding to 7956 Hz (-0.55%)

Analysing stream from 192.168.1.1 port 22808 to 192.168.1.254 port 17562 SSRC = 0x4D0618E7

Pack	Sequen	Delta(r	Filtered Jitte	Skew(ms)	IP BW(k	Mark	Status
14	54679	0.00	0.00	0.00	1.60	SET	[Ok]
16	54680	20.16	0.01	-0.16	3.20		[Ok]
18	54681	20.00	0.01	-0.16	4.80		[Ok]
20	54682	20.00	0.01	-0.16	6.40		[Ok]
22	54683	20.00	0.01	-0.16	8.00		[Ok]
24	54684	20.00	0.01	-0.16	9.60		[Ok]
26	54685	20.00	0.01	-0.16	11.20		[Ok]
28	54686	20.00	0.01	-0.16	12.80		[Ok]

Max delta = 20.16 ms at packet no. 16
Max jitter = 0.04 ms. Mean jitter = 0.01 ms.
Total RTP packets = 1220 (expected 1220) Lost RTP packets = 0 (0.00%) Sequence errors = 0
Duration 24.53 s (-139 ms clock drift, corresponding to 7955 Hz (-0.57%)

Figure 4: Jitter and Packet Lost between Cisco and Asterisk

The figure 4 above is the Asterisk RTP Stream to Cisco. The lower figure is the reverse direction. VoIP is good if jitter <20 ms. Here, through the use of only for voice network, we found the jitter <0.04 ms, and then there is no packet lost. The value of the maximum jitter always different because we do not know which package will transmit during simulation. If there is packet transmitted burst, such as ARP or DNS request, it will add a small time variation.

7.0 ASTERISK SERVER RELIABILITY

To clearly find out the need for Asterisk server resource, we will make the conference simulation coding µLaw G711 and G729a. I used the mobile Intel T5870 2GHz CPU and 2GB memory as the server. The coding G729a is proprietary, but for education or research we can use the free algorithm G729 / G723 (http://asterisk.hosting.lv).
We will measure the CPU load, used memory and throughput. Softwares for measuring are bwm-ng and top. top [14] is a standard linux software to see the system load. With top we can see the percentage of CPU usage and total memory, how much is unoccupied and lists of applications that uses these resources.
bwm-ng [15] is a software used to monitor the flow. We must first install the software because it is not the standard Linux software. It can monitor each network interface installed in the server and each direction (transmitted and received). With bwm-ng, we can save the results in the file and analysis. However, top does not allow us to save the file. I must then create a simple script to put the result of the top to file.

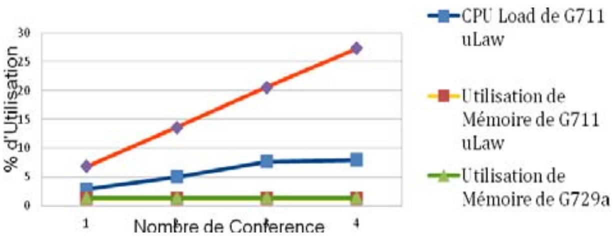

Figure 5: Comparison CPU and Memory Usage G711 and G729a

Within the results, we can see the requirement of Asterisk server resources. G 711 is a simple coding. For 1 Conference with 3 members, it needs 2.87% of CPU resources. For two conferences, each conference has three members, the CPU load increases become 5%. If we add the conference, loading is somewhat linear. However, memory usage is fixed, 1.3%. This is because Asterisk has already reserved the memory for every operation.

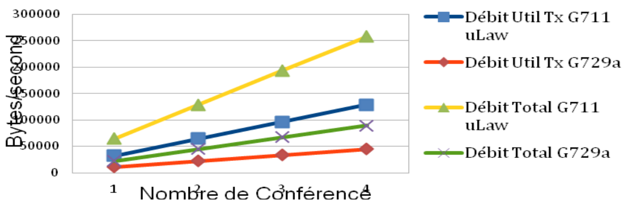

Figure 6: Comparison of Bandwidth Usage G711 vs G729a

The flow Tx and Rx are almost the same. Because the server received the packet of 3 posts in 3 positions transmitted to the well. In one conference, for example, flow about 32 KB / s. So each post used 32/3 = 10.6 KB / s or 85 kbps. Despite G711 is 64 kbps coding theory, the ethernet network adds the head that will add the flow. The number of conference increase linearly increases throughput.
G729 is a complex coding. But the annex, G729a, can reduce the complexity of calculation, however, it needs still more than the G711 resources. We can see, for 1 conference, the CPU load is 6.83%. And for two conferences, it is 13.57%. It is also as straight G711, so it's easy to predict the maximum number of conference server. The memory usage was not significantly different.
The coding rate is about 11 kbps G729a. So each post used 11/3 = 3.6 KB / s or 29 kbps. It's almost a third of G711. And the same with G711, the increased number of conference linearly increases throughput.

8.0 CONFERENCE TERMINAL

To enable Control Center dynamically choose the conference, it is necessary to add the DTMF signaling. I put the conference room number for each communication between Control Center and remote stations. 9000 Control Center A - Remote Station AA, 9001 Control Center A - Remote Station AB, 9002 Control Center A - Remote Station AC, 8000 Control Center B - Remote Station BA etc.
If the operator wants Control Center C contact Remote Station AB, he easily dial the conference room number 9002. If the operator of Control Center A wants to join communication with the Remote Station BA, he can dial 8000. The call number is transmitted DTMF to Asterisk. So in each CROSS there is 1 device dedicated to follow the conference.

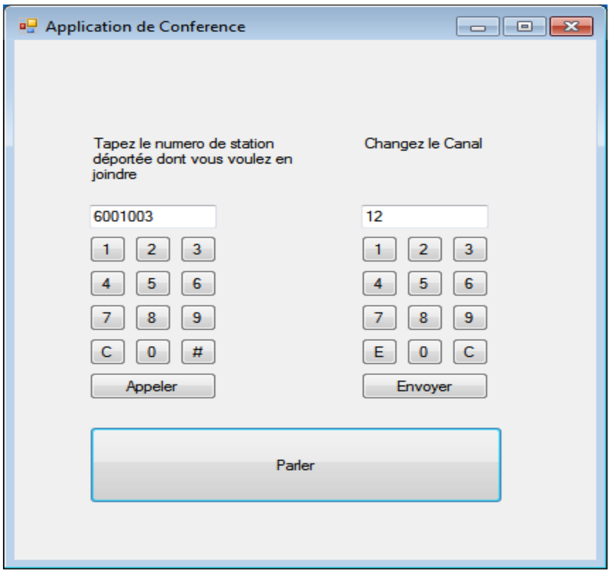

Figure 7: Interface of Conference Terminal

The application interface is like the figure 5 above. To the left are the buttons to call the Asterisk server and ask him to join the conference. For example, 6001003 is the conference room between Control Station A and remote station AB. To the right are the buttons to change the channel remote station.
When we push "Send" button, the application will generate the CCIR code. For Control Center, the channel change rule is: the transmit frequency which is correspondence with 110xx symbol, where xx is the channel number. For example, channel 12, the generated code is the CCIR 11012. If there are two like symbols that are adjacent, the second symbol is changed with symbol E. Therefore, the generated code is 1E012. Each symbol lasted 100 ms.

Table 2: CCIR Code

CCIR symbole	Tone Fréquence	CCIR symbole 2	Tone Fréquence 2
1	1124 Hz	9	1860 Hz
2	1197 Hz	0	1981 Hz
3	1275 Hz	A	2400 Hz
4	1358 Hz	B	930 Hz
5	1446 Hz	C	2247 Hz
6	1540 Hz	D	991 Hz
7	1640 Hz	E	2110 Hz
8	1747 Hz	F	-

In the lower the button to the frequency 2800 Hz. Push the button, and then talk. Push again to stop the emission of the frequency 2800 Hz.

9.0 CONCLUSION

Communication between Cisco and Asterisk is the same with communication between softphone and Asterisk. The delay is less than 150 ms round trip, jitter less than 20 ms and packet lost less than 0.5%. They comply with the ITU G .114 for good voice quality. During conference with 3 participants, there is no degradation of voice quality served.
With the comparison between G711 and G729a and simulation to 4 simultaneous conferences, the results of the load CPU, memory and speed are good and consistent. We can expect the need for resources if we set up the conference.
To keep compatibility with legacy equipment (radio communication with ships), we have to create application like conference terminal explained above. Asterisk and Cisco doesn’t have feature of CCIR.
I propose to distribute the Asterisk server in all Control Station sites and add redundancy. Each remote station has connection to Control Center using MPLS VPN. Every Control Center has MPLS VPN connection to another Control Center. This configuration will save the VPN connection. We can also consider using the hub and spoke configuration for connection among Control Center.

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