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# Voip Protocols

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*This article focuses on existing technologies in telecommunications but also on a comparative analysis of VoIP protocols. There will also be listed ways to extend networks and processes that contribute to the steady operation of the network as a set SLA with the support of a large number of simultaneous connections*

***Keywords:*** *VOIP, H.323, Q.931, VRRP*

## Introduction

**Extending a H.323 Standard Based Network**. H.323 standard describes a series of functional components which can be implemented separately, in different pieces of equipment, or can be grouped in one piece of equipment with multiple tasks. We can remember characteristics of the mentioned standards:

The possibility of realizing videoconferences: MC (Multipoint Controller) – ensures the necessary functions for making a videoconference between 2 or more terminal points. The MC establishes a control channel H.245 with each attendee at the conference. Through this there are exchanges of pieces of information regarding the type of conference (centralized/

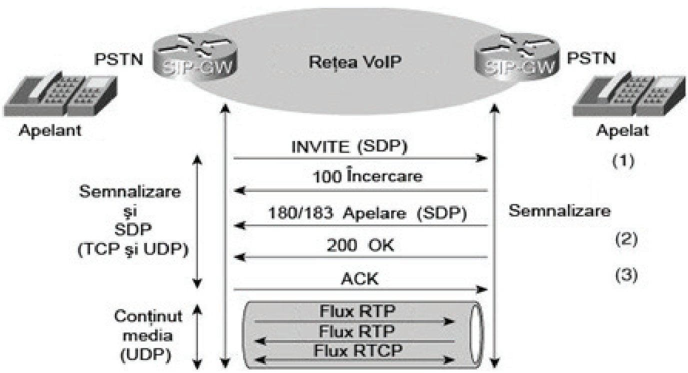
decentralized). The MC is not a proper equipment, it is embedded in a terminal point (final or gateway), gatekeeper or MCU.

It processes (mixing and commuting) audio and video signals for all attendees. The MP (Multipoint Processor) – adds functionality to the conference. Like MC, the MP is embedded in the MCU. The MCU is an independent piece of equipment, shaped as a terminal equipment, that ensures a multi-terminal videoconference, embedding an MC and plenty of MC (or none).

Extension. A H.323 network can be extended easier if used a gatekeeper in its infrastructure. In this subchapter we will analyze the functionality of the network that includes a gatekeeper, but also keep track of its behavior in the absence of the mentioned equipment.

The main procedure for initializing a H.323 call presupposes a high number of exchanges of information between gateway-source and gateway-destination. Following the Fast Connect procedure, the number of exchange processes will be reduced, permitting negotiation for the capabilities and the attributes of the logical channel in a single input-output message between terminal points.

**Figura 1:** Fast Connect




As it can be seen in Fig. 1, the procedure Fast Connect implies effecting the sequent steps:

- The origin of the gateway initializes a session of H.225 towards destination-gateway through TCP 1720 port.

- The initializing process of the call, based on the Q.931 protocol, creates a combined logical through which it will be realized not only by signalizing between terminal points, but signalizing for the H.245 specified control function. The negotiation of the capabilities and proprieties of the logical channel is transmitted with the signalizing process of Q. 931.

- Taking into consideration the proprieties of the logical channel, the RTP sessions will be open.

- The terminal points will realize the exchange of multimedia information through RTP sessions.

## Methods and strategies for protecting the errors

The vulnerability of every environment that depends on a common control equipment is proportioned directly to the probability of that control equipment to fail. In the traditional telephone network, the safety against all damages is improved by adding control equipment.

A strategy to solve the problem is to double the number of the equipment with an important role in the network. This pricey onset is usually replaced with another solution that ensures the redundancy of the network, more exactly, one substitute piece is ready to replace a damaged one. In both cases, the strategy means having a back-up solution for the key-components of the network. I will put forward some methods of protection against errors that contributes at a permanent functionality of the network:

• **Adding more gatekeeper equipment to the network means reducing the risk of lack of access to the gatekeeper**. Every gatekeeper added creates an unique H.323 zone. Because every terminal point is associated with one gatekeeper, in a single H.323 zone the terminal points need to be configured so as to identify only one of the multiple gatekeepers that work at a certain moment in time. To solve this problem, the gateways need to be configured with a list of functional gateways so as to make a systemized identification.

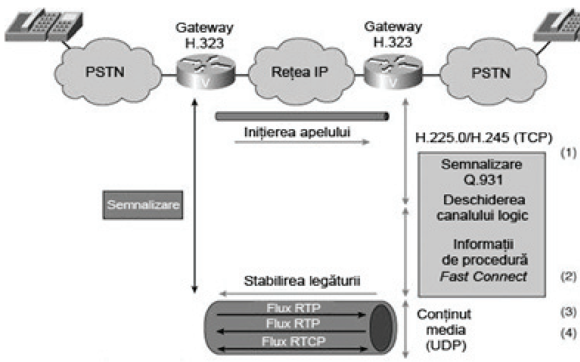
• **HSRP** (Hot Standby Router Protocol) allows setting only one IP address for 2 gatekeeper equipments, and also allows access to both equipments on the same local network (LAN). However, the access to the network is done one at a time, one gatekeeper is active permanently, the other being the backup solution in case of a malfunction. Terminal equipment are is configured to recognized the name of the gatekeeper which can be translated in an IP address recurring to DNS [105] solution.

• **VRRP** (Virtual Router Redundancy Protocol) allows many gatekeeper equipments the simultaneous access to the network, on a single bond and using a single IP address. One of the equipment acts like a master, and the other are configured to act as backup equipment. VRRP is defined by IETF as RFC 3768.

• **Configuring more gatekeeper equipment with the same prefix.** To assure a permanent function of the network at a gateway level means configuring more gateways with a bond to the same stand from SCN [59] network.

• **Adding Fast Connect procedure.** This procedure will reduce the number of exchange processes, allowing negotiation for the capabilities and the attributes of the logical channel with one single input-output message between terminal points.

**Figure 2:** Fast Connect Procedure




As we can observe in Fig. 2, the procedure Fast Connect implies following the next steps:

1. The original gateway of the call initializes a H.225.0 session towards the declinational gateway through TCP 1720 port.

2. The process to initiate the call, having as basis Q.931 protocol, creates a logical combined channel through which it would be obtained not only signalization among terminal points, but it would be attained the specific control function H.245. Negotiation of the capabilities and the characteristics of the logical channel are send with the signalization process Q.931.

3. Taking into consideration the characteristics of the logical channel, the RTP sessions will be open.

4. Terminal points will make the exchange of multimedia information

through RTP sessions.

## Comparative Chart

In a general model, the components of signalization and control of the call are classified in common control components and terminal points. Common control components are making sure of the following services: administration of the call, condition of the call, management of the addresses and control of the access to the network. In chart 2.1 there are identified the components of the general model and functions executed by them for each of the 3 protocols:

The specifications of the SIP standard aren't carrying out the specific aspects of the call, as they are described in H.323 standard. The goal of SIP standard is to create, modify and fulfill sessions between different applications, regardless of the type of media content or the functions of that specific application. The session can be a phone conversation between 2 or more users, multimedia conferences or sessions of interactive games. SIP cannot define the type of session; it can only deal with its management. For this, SIP has the sequent basic functions:

• Localization of the users by translating their SIP address in IP address.

• Negotiation of the capabilities between all participants at a session.

• Modification of the session's parameters in the moment of the actual call.

• Realization of the processes of connecting and ending a call for all participants at the session.

To define other aspects related to VoIP and multimedia sessions, SIP uses other protocols developed by IETF. For example, URL is used for addressing, DNS for resolution of addresses and TRIP (Telephony Routing over IP) for routing the call in the network. SIP uses MIME (Multipurpose Internet Mail Extension) to describe the content of its messages.

SIP has the function of user's mobility; the user can be identified in a session after a number/own personal name. Also, there are other specific facilities of the intelligent network (IN – Intelligent Network) offered. Thus, the phone operators can implement fast new services in the equipment from outside the commutation center spot.

Because of the IETF efforts, there were added a lot of extensions at the SIP protocol in order to foster operation with classical voice networks. However, the main reason to use the standard would still be creating a software development environment for communicational platforms for the next generations. [27, 31]

## Advantages

**H.323** was for a long period of time the only viable solution for signalizing and controls the call in a VoIP environment. As a consequence, H.323 equipment is frequently found in VoIP networks.
When the protocol is well implemented, the network (with a distributed architecture) can be easily extended by adding new branches. Also, the functionality of the network can be improved through adding new services. H.323 is a model of control suitable to be implemented inside companies, the centralized control done by the gatekeeper facilitating the operation, the management and the maintenance of the network.

• Identifying the caller (caller ID) – this function is due to information exchanged at FXO ports and signalization on an associated channel of T1 line;

• Interoperability – H.323 is uses at a high scale, pulled together without any problems with the applications and the equipment of other producers. Because all the equipment needs to bare all primary protocols within H.323 standard, the fact of using one equipment or another is independent from the version of the standard.

• The detailed control of the call – H.323 allows a detailed research of the call from and towards the gateway, like analyzing the number pressed, equally distributing the traffic through different ways of communication or directing it towards a new route of the call.

• Integrating different technologies in the network – systems can be integrated in H.323 network, being based on classical telephone system or ISDN lines. H.323 can hold much more types of TDM interfaces and signalization than MGCP.

• Pylon for different media content – H.323 can be used for services of voice and videoconferences, also for data traffic.

• Pylon for signalization protocol NFAS (Non-Facility Associated Signaling) – this protocol allows plenty of ISDN PRI lines through one single signalization channel type D, having at basis a lot more free channels.

• Gatekeeper H.323 – a gateway can address to a gatekeeper to perform the control functions of the call and the resolution of the addresses.

SIP operates independent from the type of the session or the media content, this making it easy to be used.

1. It is an opened standard, having the support of plenty producers which implement SIP in their equipment. The applications can be developed congruent to the subsequent usage of the equipment.

2. SIP messages are text types, making easier to identify and resolve possible problems.

3. SIP allows plenty of users with different capacities to operate simultaneously. For example, in a conference where there are users with video capacities and users only with audio capacities, the ones with video capacities could carry on the session using both media capacities. They will not feel the obligation to give away their video capacity, participating only with the audio side, this fact happening in other protocols.

• MGCP. The easy way to configure equipment – the gateway takes in hand part of the configuration information from the call agent. A special care needs to be given to the situation when the gateway needs to ensure the fallback function, situation that implies configuration of the gateway with H.323 protocol.

• Optimal management for network functioning – because the call agent has the necessary "intelligence" to control the call, the management of plenty of gateways is centralized.

• Survival of the call – the gateway keeps the calls in an active state through analogical and digital lines (interfaces type T1 CAS). This happens in the transition period towards a backup call agent or until SRST process begins.

• Coding of the voice traffic – MGCP was the first gateway control protocol that ensures coding the voice traffic using SRTP (Secure RTP).

## Restrictions

**H.323** H.323 protocol has a series of disadvantages too:

• Configuration – To configure a gateway is a more complex process than in case of the MGCP protocol because it means introducing a numbering scheme. Using a gatekeeper would reduce the complexity of the configuration process.

• Lack of a centralized numbering scheme – If the numbering scheme needs certain changes, then all the gateways from the network would need to be reconfigured. Using a gatekeeper would help in a certain way.

• Survival of the call – H.323 basic configuration doesn't have this function. If the connection with the gatekeeper are lost, then all the calls would be interrupted. For using SRST technology – Survivable Remote Site Telephony, all the active calls would be taken over after reestablishing the connection with the gatekeeper and that specific area.

**SIP.** Processing text messages implies a an additional charge of the gateways. The router has to translate the text in an understandable language, and the code for this operation needs to be included in the operating system of the equipment.

• SIP is a recently appeared standard (2002), part of its functions still being developed. For din reason, a large amount of the producers prefer implementing a personal choice of the standard in equipments.

**MGCP.** Depending on the call agent – gateways don't have the necessarily "intelligence" to control a call. For this reason, they are ordered by the call agent. To ensure permanent function of the network, the network administrator needs to ensure the redundancy of the network through different methods (SRST or H.323 fallback).

• Calls from PRI and BRI interface lines are ceased – when the connection towards the call agent is interrupted, calls from the mentioned lines are not kept in an active state.

• Calls for the fax – although MGCP protocol has its own different methods for treating call through fax, the call agent cannot take negotiation and signalization procedures for this type of call.

## Conclusions

As a conclusion, the expectations are high for SIP standard. It is seen as a software platform that is involved in developing multimedia communications, permitting fast communicational sessions, in any moment and any circumstances. Still, SIP is for now a protocol for control of the call, with its pros and cons.

Although it is permanently developed new extensions that could allow networks based on SIP protocol to work together with classical telephone networks, the main reason of developing the protocol is to create a software platform for the future generation networks (NGN – Next Generation Networks). SIP is a multimedia protocol that uses an architecture and messages similar to those encountered in the specific Internet applications. By using a type URL method of addressing and text messages, SIP benefits from the advantages offered by the Internet network at the implementation of VoIP applications in the networks.

**H.323**. When H.323 protocol is well implemented, the network (with a distributed architecture) can be easily extended by adding new branches. Also, the functionality of the network can be improved by adding new services. H.323 is a control model suitable to be implemented inside the companies, centralized control made by the gatekeeper facilitating the operation, the management and the maintenance of the network.

**MGCP.** MGCP protocol describes an architecture where the control of the call and the services offered (operation, management and maintenance of the network) are grouped in a main entity, VoIP network having it as base of development. We can say that MGCP architecture is resembled to the architecture and the services of the telephone's public network (PSTN). It's a flexible protocol regarding adding new functions to the call agent entity. MGCP is suitable for service providers because their network needs a centralized check. MGCP suggests a series of mechanisms of interconnection with the other VoIP networks.

## References

[1] G. Gruman, E. Knorr, What cloud computing really means. *InfoWorld,* (2009, May).[Online].Available:http://www.infoworld.com/d/cloudcomputing/what-cloud-computing reallymeans-031

[2] L. SIEGELE, Let it rise: A survey of corporate IT. *The Economist,* (Oct., 2008).

[3] P. Watson, P. Lord, F. Gibson, Panayiotis Periorellis, and Georgios Pitsilis. *Cloud computing for e-science with carmen*, (2008), pp. 1–5.

[4] A. Ghencea, V. Gaucan, D. Pirvu, "Distributed Systems and Web Technologies", *Journal of Knowledge Management, Economics and Information Technology, Vol. I, Issue 5, August 2011*

[5] M.-E. Begin, An egee comparative study: Grids and clouds – evolution or revolution. *EGEE III project Report*, vol. 30 (2008).

[6] B. Rochwerger, D. Breitgand E. Levy, A. Galis, K. Nagin, I. M. Llorente, R. Montero, Y. Wolfsthal, E. Elmroth J. Caceres, M.Benyehuda, W. Emmerich, F. Galan, The Reservoir model and architecture for open federated cloud computing. *IBM Journal of Research*

*and Development*, vol. 53, no. 4 (July, 2009), pp. 1–11.
[8]“Implementing QoS Solutions for H.323 Videoconferencing over IP”, *Cisco Systems Technical Whitepaper Document Id: 21662*, 2007.
[9]. P. Calyam, M. Haffner, E. Ekici, C.-G. Lee, “Measuring Interaction QoE in Internet Videoconferencing”, *Proc. of IFIP/IEEE MMNS*, 2007.
[10]. S. Winkler, “Digital Video Quality: Vision Models and Metrics”, J*ohn Wiley and Sons Publication*, 2005.