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Voice and Data Integration and Interoperability. SIP-H.323 Protocols Translation Simulator

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Abstract – The VoIP protocols have now a very fast evolution and diversity. The paper presents a solution for VoIP service through multiple interconnected IP networks, each network using a different VoIP protocol: H.323 or SIP. Both H.323 and SIP are widely implemented as default VoIP protocols in data networks. Both protocols can be used for setting up Internet multimedia conferences and telephone calls. In order to achieve universal connectivity, interoperability between the two protocols is highly requested. That interoperability manages to a full protocol translation from one network to another and supports unified Voice service through the all networks. The paper presents such a H.323-SIP interconnection simulation program.
Keywords: VoIP, SIP, H.323

I. INTRODUCTION

Today, the communication networks have became the key of any information system. The TCP/IP protocols stack was developed for the very internetworking purpose. It allows the fast development of Internet and private IP networks (fig.1).

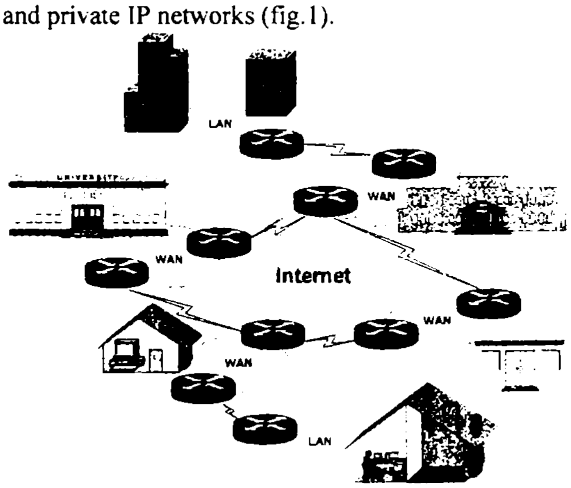

Fig. 1. Internet connectivity

The highest adaptability of IP technology to almost any kind of physical network makes possible the true convergence of services. A user accesses the data network and voice network in the same way in a converged network environment.

The migration of voice traffic from PSTN to IP networks and the deployment of next-generation broadband services based on the low cost of VoIP calls and the higher flexibility of IP technology became a trend in communications world (fig.2).

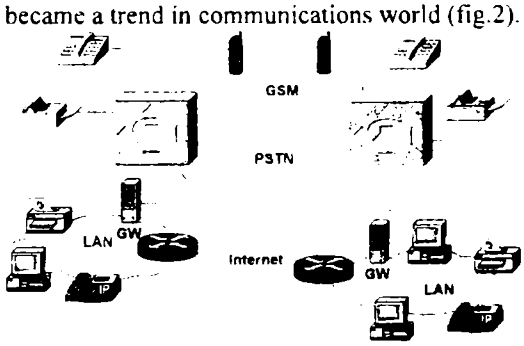

Fig. 2. Universal services on next generation network

The VoIP protocols have now a very fast evolution and diversity. Following this consideration, the paper presents a solution for VoIP service through multiple interconnected IP networks, each network using a different VoIP protocol (H.323/SIP). Our contribution to that domain is an H.323-SIP protocol translation program, developed as an analyzer tool for H.323 to SIP network migration/extension.

II. THE H.323 STANDARD

H.323 is the most important family of ITU-T standards for real time information transfer, through packet based networks, including Internet (LAN, MAN, WAN, enterprise). H.323 standard is a basic technology in real time audio, video and data transfer, over packet based networks. It specifies components, protocols and procedures by providing multimedia communication in packet based networks.

H.323 can be applied to a variety of systems-voice only (IP telephony); voice and video (video telephony); voice and data; audio, video and data. H.323 can also be used for multipoint multimedia communications.

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H.323 is considered to be an "umbrella" protocol, which defines all aspects concerning transmission, starting with call initiation and finishing with communications facilities provided by the resources available within the network. The H.323 functionality and independency is based on the following protocols and algorithms (fig.3 and 4):

- audio codecs
- video codecs
- H.225 registration, admission, and status (RAS)
- H.225 call signaling
- H.245 control signaling
- Real-time Transfer Protocol (RTP)
- Real-time Control Protocol (RTCP).

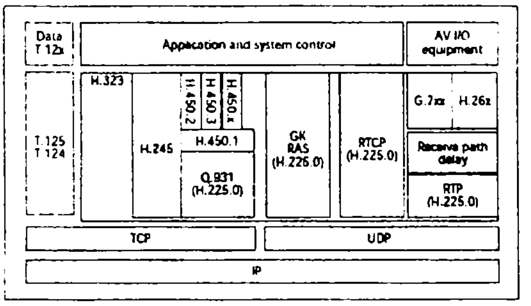

Fig. 3. H.323 protocol stack

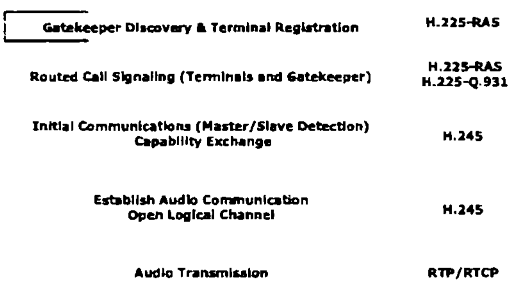

Fig. 4. H.323 signaling flow

In order to provides point-to-point or point-to-multipoint multimedia communications services H.323 standard defines four types of components (fig.5):

- terminals
- gateways
- gatekeepers
- multipoint control units (MCU).

A. H.323 terminal

The H.323 terminal can be a PC or stand-alone device, running an H.323 stack and multimedia applications. It uses H.245 for exchanging terminal capabilities and creation of media channels. Also, this terminal makes use of H.225 for call signaling and call setup, and RAS for registration and other admission control with gatekeeper. In the same time H.323 terminal uses RTP/RTCP for audio and video streams.

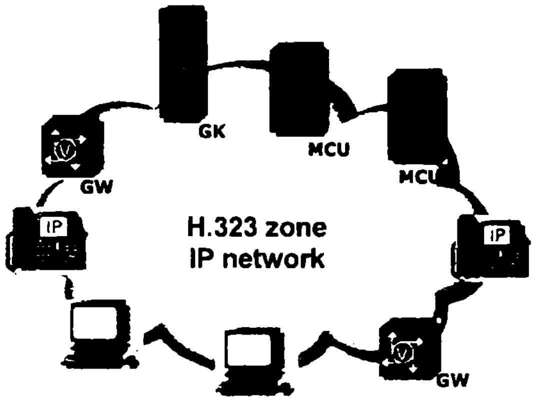

Fig. 5. H.323 network

B. H.323 Gateway

An H.323 gateway connects an H.323 network and a non-H.323 network (ex. SIP, PSTN). It performs protocol translations for call setup and release Converts media formats and allows information transfer between H.323 network and non-H.323 networks.

Terminals communicate with gateway using the H.245 control signalling protocol and H.225 call-signaling protocol. The gateway translates protocol's information in a transparent fashion for both communication end-points.

The H.323 gateway also performs call setup and call clearing on both the H.323-network side and other network side.

C. H.323 Gatekeeper

The H.323 gatekeeper represents the main component of H.323 network. It performs addressing, authorization and authenticity of terminals and gateways. The gatekeeper handles bandwidth management, accounting, billing, charging and also performs call-routing.

D. H.323 Multipoint control unit (MCU)

The H.323 MCU provides support for conferences of three or more H.323 terminals. All terminals participating in the conference establish a connection with the MCU.

Being a protocol used for distributed architectures, H.323 allows companies to build scalable, resilient and redundant wide area networks. It provides mechanisms to interconnect with other VoIP networks and accepts network intelligence at terminal level as well as central devices (gatekeeper).

H.323 includes facilities like "packet-based fax", gatekeeper-gatekeeper communications and mechanisms for fast connections. Due to its availability right at the beginnings of VoIP implementation and to its adaptability to IP requirements, H.323 is now the most widely used signaling and call control VoIP protocol.

III. THE SIP STANDARD

The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance e-learning, Internet telephony and similar applications.

SIP has been designed as a multimedia protocol using a distributed architecture with universal resource location (URL) for text-based messages, trying to take advantage of the Internet model for creating VoIP netwoks and applications (fig.6).

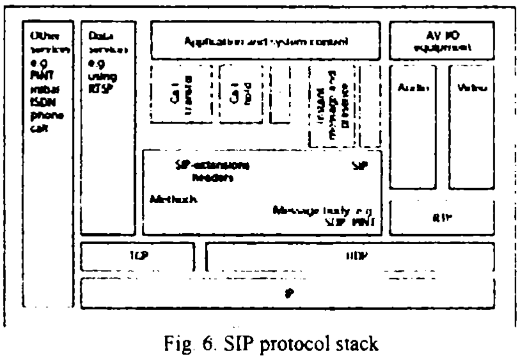

Fig. 6. SIP protocol stack

The SIP standard defines four entity types, as shown in fig.7:

- user agent (UA)
- proxy server (Proxy)
- registration server (Registrar)
- Redirect server (Redirect).

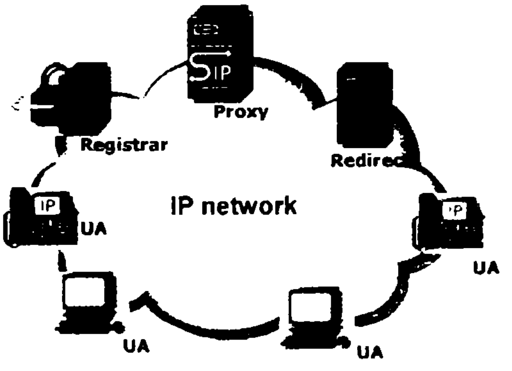

Fig. 7. SIP network

A. SIP user agent (UA)

The SIP UA is an endpoint device that terminates the SIP signaling (fig.7).It can be a client (UAC) that initiates requests; a server (UAS) that responds to request or a combination of both.

B. SIP proxy

The SIP proxy is a device in the signaling path between to different UAs, that routes requests to their destinations (fig.7 and 8).

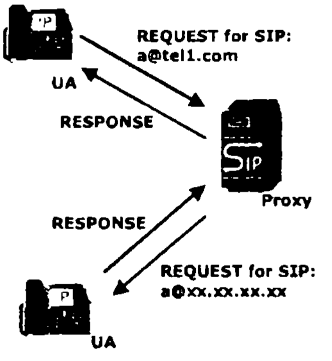

Fig. 8. SIP proxy server

It can add parameters to the requests or rejects requests. Also, it may not initiate requests or respond positively to any request.

C. SIP registrar

The SIP registrar is a Specialized UAS that handles REGISTER requests (fig.7 and 9).

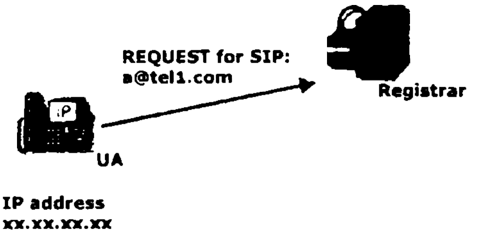

Fig. 9. SIP Registrar

D. SIP redirect server

Named rerouting server, too, SIP redirect server is an UAS that responds to requests by redirecting them to another device (fig.7 and 10).

As a protocol used in a distributed architecture, SIP allows companies to build scalable, resilient and redundant large scale newtorks. The protocol provides mechanisms to interconnect with other VoIP networks in order to add intelligence and new options to each of the terminals, SIP proxy servers and rerouting servers.

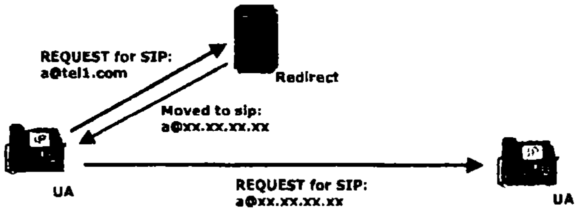

Fig. 10. SIP Redirect server

SIP communication is made up of me__age_ _ha_ are sent between the devices using UDP, TCP, or another :ranspor: pro.ocol. A single SIP reques: _nd _ll i.s responses form a SIP transaction, as shown in fig.11.

```
INVITE sip:7170@iptel.org
SIP/2.0
Via: SIP/2.0/UDP
195.37.77.100:5040;rport
Max-Forwards: 10
From: "jiri"
<sip:jiri@iptel.org>;tag=76ff
7a07-c091-4192-84a0-
d56e91fe104f
To: <sip:jiri@bat.iptel.org>
Call-ID: d10815e0-bf17-4afa-
8412-
d9130a793d96@213.20.128.35
CSeq: 2 INVITE
Contact:
<sip:213.20.128.35:9315>
User-Agent: Windows RTC/1.0
Proxy-Authorization: Digest
username="jiri",
realm="iptel.org",
  algorithm="MD5",
uri="sip:jiri@bat.iptel.org",
```

Fig. 11. SIP message example

A SIP dialog means a persistent link between two devices that is used to associate transactions. A call contains multiple dialogs. SIP message contains a call identifier field (Call-ID) that is used to link the dialogs and transaction into an application-level concept of a call.

IV. SIP-H.323 INTERCONNECTION

Both H.323 and SIP are widely implemented as default VoIP protocols in data networks. Bboth protocols can be used for setting up Internet multimedia conferences and telephone calls. In order to achieve universal connectivity, interoperability between the two protocols is highly requested. That interoperability manages to a full protocol translation from one network to another and supports unified Voice service through the all networks.

SIP-H.323 interconnection requires a transparent support for the signal and for session description between the SIP and H.323 entities. A server which performs this translation is called SIP-H.323 Signalling Gateway (fig.12).

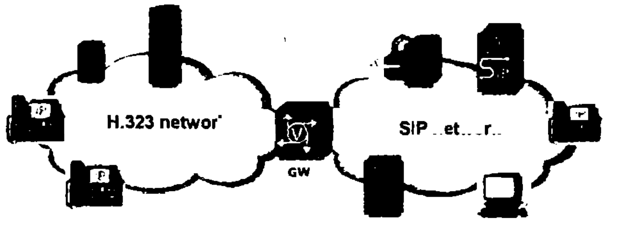

Fig. 12. H.323-SIP interconnection

According to the type of information shared between the H.323 or SIP and GW elements, it is possible to create a variety of architectures to achieve transparent address conversion and establish a call.

A. H.323 signalling

Under H.323, a call between two terminals („A" and „B") is established as follows:

- both terminals must be registrated before any communication begins;
- „A" sends a gatekeeper admission request;
- the gatekeeper finds „B's" IP address and sends it to „A";
- „A" establishes a TCP connection to „B" and sends a Q.931 SETUP message;
- „B" answers with a Q.931 CONNECT message;
- once the first stage of the Q.931 signal is completed, H.245 takes over the signalling control.

B. SIP signaling

SIP initiates a call through a INVITE message and an answer from the called party. Both the invitation and the answer contain a session description which indicates the terminal capacity. Proxy and rerouting servers are responsible for the parties' user names and IP addresses translation.

C. *SIP-H.323 signaling* (fig.13, 14 and 15)

Both H.323 and SIP are widely implemented as default VoIP protocols in data networks. Bboth protocols can be used for setting up Internet multimedia conferences and telephone calls. In order to achieve universal connectivity, interoperability between the two protocols is highly requested. That interoperability manages to a full protocol translation from one network to another and supports unified Voice service through the all networks.

Generally, three kinds of informations are needed to establish a call between two points:

- signalling destination address;
- local and distant capacities;
- local and distant media transport addresses for which the destination points can receive media packets.

In case of H.323, this information is stretched over various call stages, while SIP converts it in a Invite and Answer messages.

There is a direct translation from a SIP call to a H.323 call. The GW takes the three information items contained in the SIP invitation and splits them over the multiple stages of establishing a H.323 call. For the opposite direction, from H.323 to SIP, the various stages of a H.323 call have to be united in only one SIP INVITE message.

The SIP-H.323 conversion must also solve the user authentification problem. User registration means keeping track of user names, phone numbers or different other ID data, like email or network addresses. Allowing users to be found by independent location identifiers, user registration allows personal mobility.

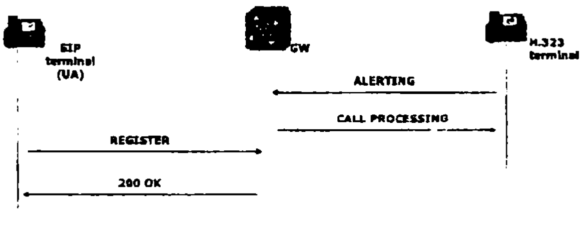

Fig. 13. Registration

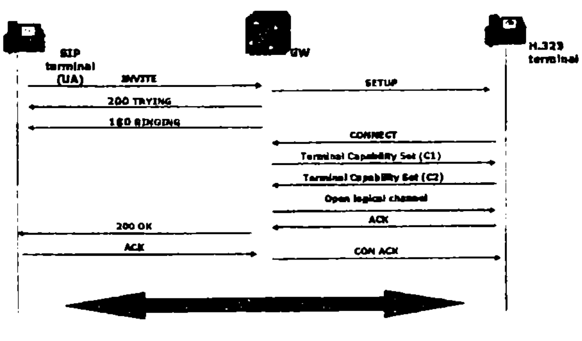

Fig. 14. Call initiation and data transfer

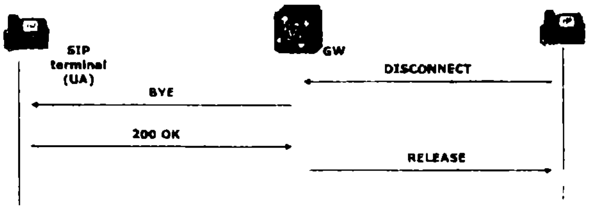

Fig. 15. Call termination

V. SIP-H.323 INTERCONNECTION SIMULATION PROGRAM

The simulator has been developed for an IP network which has the VoIP service implemented according to the SIP standard. To be fully operational, the VoIP service under SIP must ensure the interconnection with other IP networks and the compatibility with the VoIP services implemented using the H.323 standard. The interconnection is achieved through a SIP-H.323 gateway whose functioning is simulated by this program (fig.16).

The program shows communication stages between two terminals located in different VoIP networks (SIP and H.323 respectively), taking into consideration all cases that might appear. Within the simulation there are being used messages specific to the two types of protocols, the protocol conversion being done at SIP gateway or H.323 gateway level.

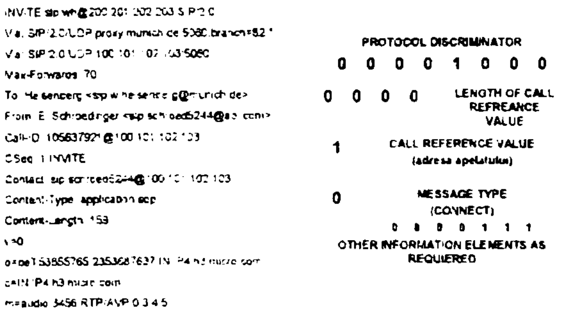

Fig. 16. SIP-H 323 message protocol translation

The program follows the three main states of a call:

- call establishing or call initiation (fig.17 and 18)
- audio data transfer (fig.19)
- call terminating (fig.20 and 21).

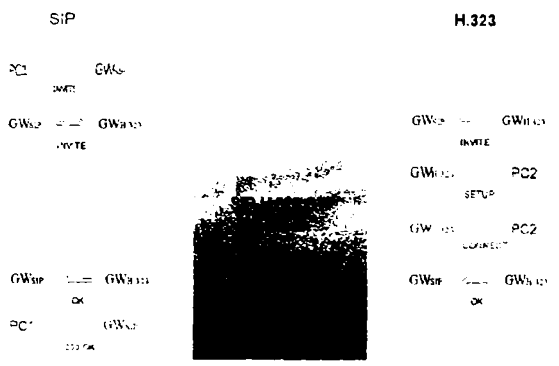

Fig. 17. Signaling messages – call establishing (initiated)

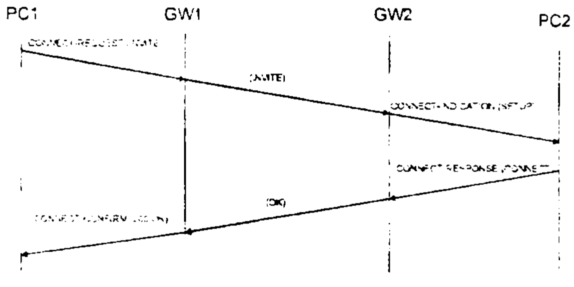

Fig. 18. Service primitives- call establishing

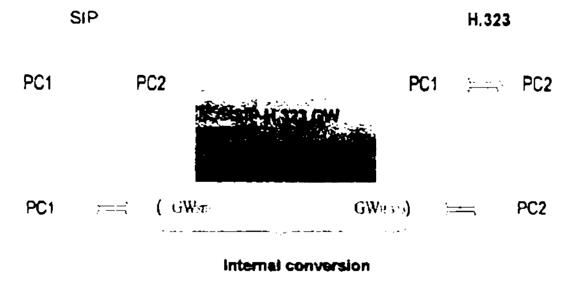

Fig. 19. Audio data transfer between the SIP terminal and H 323 terminal

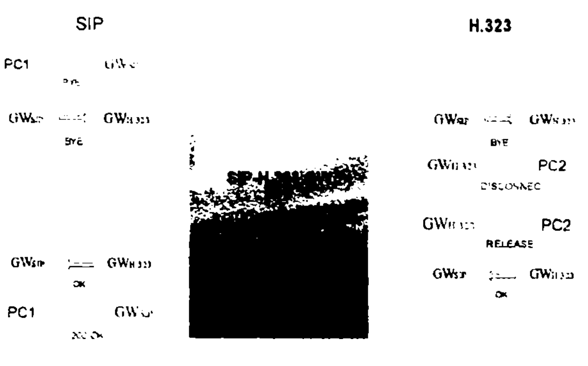

Fig. 20. Signaling messages- call terminating

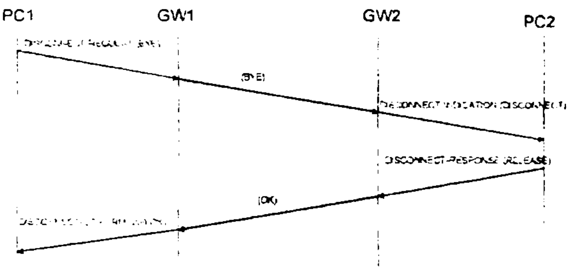

Fig. 21. Service primitives- call terminating

For each state there are several service primitives used, which are converted into messages according with each protocol.

In order to cover all steps follow by the call in an interconnected networks environment, the program respects the call state diagram (fig.22).

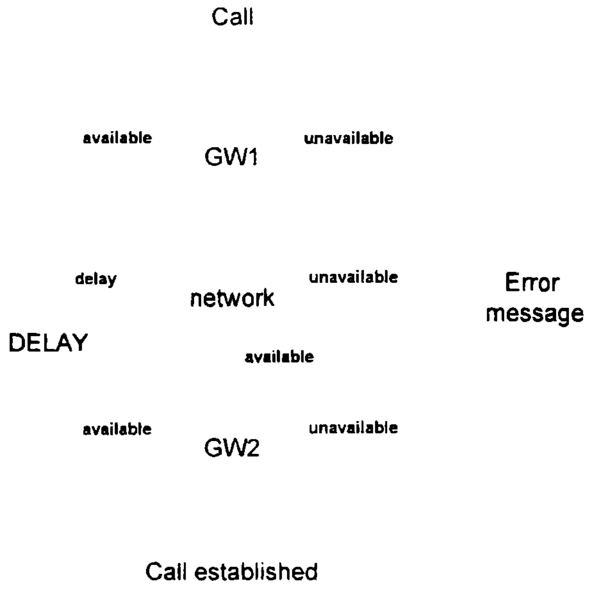

Fig. 22. Call state diagram

The program starts with user registration for both SIP and H.323 terminals. When SIP terminal wants to make a call, there are a messages exchange between it and the SIP gateway.

If all different component states are favourable to call ......a...... a..._ a.s_ ..._ u.... ..am_ c.....sp.....s .. ...e registered user name, call initiation will proceed correctly. The simulator shows all messages involved in the procedure and the H.323 user will be notified by the call (fig.23).

The cal. rejecting procedure is identi al to the call terminating procedure.

The call acception procedure is initiated only when H.323 user accepts the call.

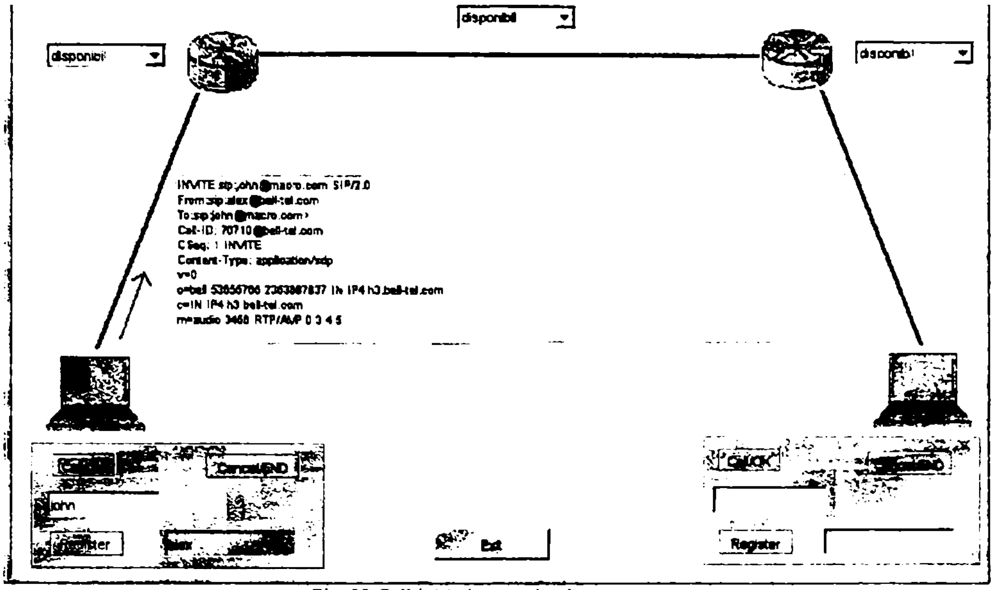

Fig. 23 Call initiation on simulator program

Connection establishing is shown through call indicators for both users and conversation can start (fig.24).

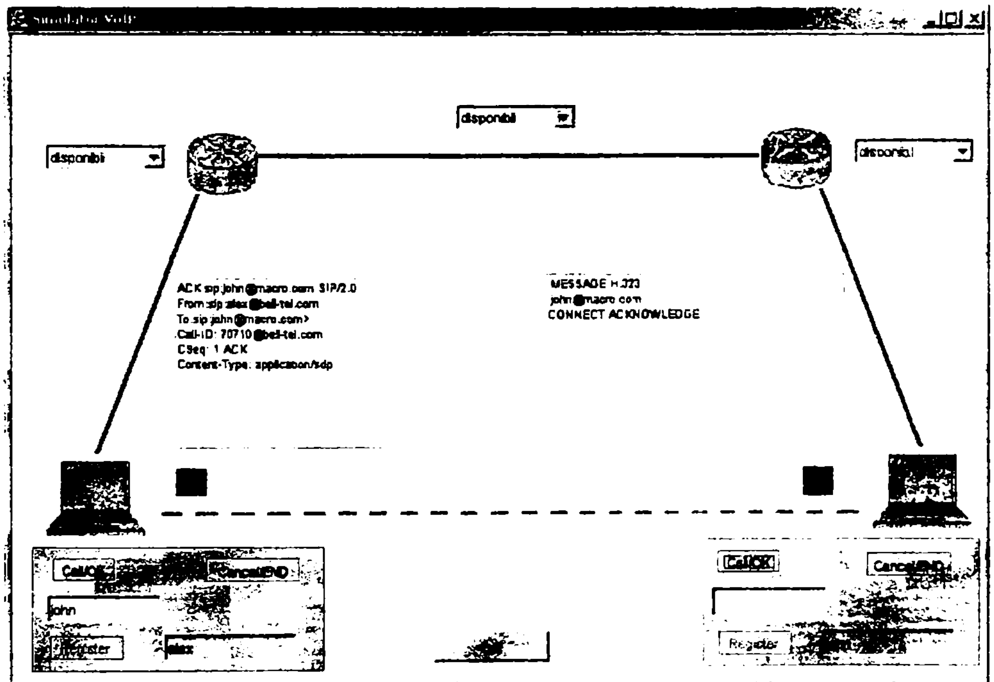

Fig. 24 Call established and communication initiated on simulator program

VI. CONCLUSION

The simulator provides a possibility to test the SIP-H.323 gateway under various working conditions, useful for a network administrator, which can thus choose the interconnection way and the transparent functions between the SIP and H.323 networks.

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