A

Seminar report

On

"IP Telephony"

Submitted in partial fulfillment of the requirement for the award of degree
of Bachelor of Technology in Computer Science

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Acknowledgement

I would like to thank respected Mr…….. and Mr. ……..for giving me such a wonderful opportunity to expand my knowledge for my own branch and giving me guidelines to present a seminar report. It helped me a lot to realize of what we study for.

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Next, I would thank Microsoft for developing such a wonderful tool like MS Word. It helped my work a lot to remain error-free.

Last but clearly not the least, I would thank The Almighty for giving me strength to complete my report on time.

Preface

I have made this report file on the topic **IP Telephony**; I have tried my best to elucidate all the relevant detail to the topic to be included in the report. While in the beginning I have tried to give a general view about this topic.

My efforts and wholehearted co-corporation of each and everyone has ended on a successful note. I express my sincere gratitude to …………..who assisting me throughout the preparation of this topic. I thank him for providing me the reinforcement, confidence and most importantly the track for the topic whenever I needed it.

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1. Introduction:

Today IP Telephony is a very powerful and economical communication options.

IP telephony is the integration and convergence of voice and data networks, services, and applications. Internet telephony uses the Internet to send audio, video and data between two or more users in the real time.

Internet Engineering Task Force (IETF) defines the IP telephony is the exchange of information primarily in the form of speech that utilizes a mechanism known as Internet Protocol.

ITU-T Study-Group 2 (SG2) issued the following explanations of the term IP telephony: "IP is an abbreviation for Internet Protocol. It is a communications protocol developed to support a packet-switched network.

The main motivation of development of IP Telephony is the cost saving & integrating new services. Internet telephony integrates a variety of services

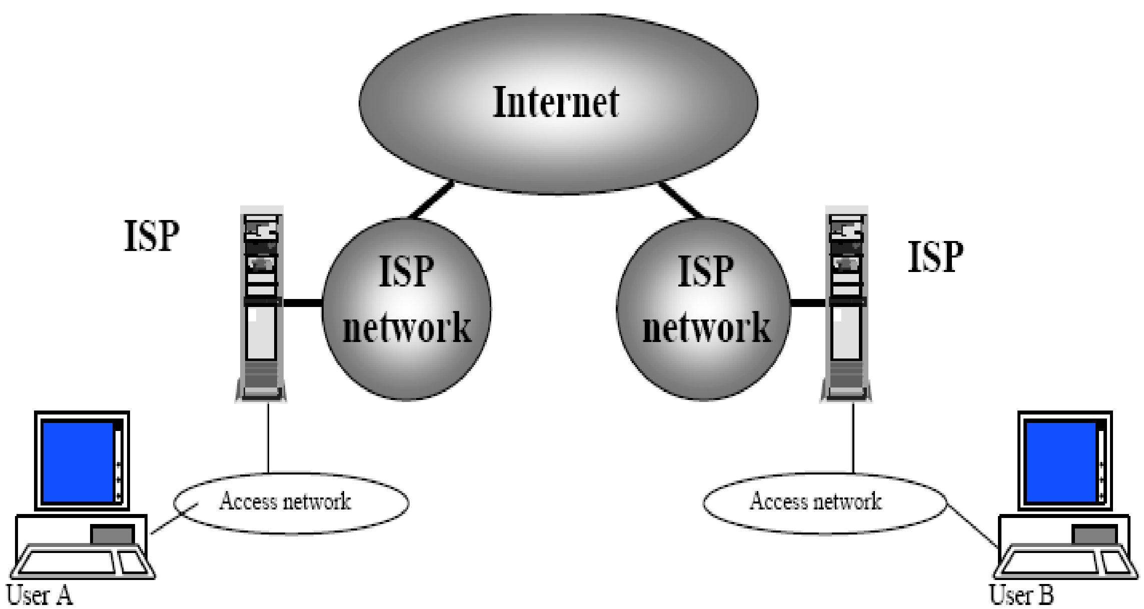
Vocaltec introduced the first Internet telephony software product in early 1995, running a multimedia PC, the Vocaltec Internet Phone. In 1996, Vocaltec announced it was working with an Intel Company (Dialogic Corporation, an Intel acquisition made in 1999) to produce the first IP telephony gateway. The technology has improved to that point where conversations are easily possible. Gateways are the key to bringing IP telephony into the mainstream. By bridging the traditional circuit-switched telephony world with the Internet. Internet telephony technology has caught the world's attention.

2. Different types of IP telephony:

There are four types IP telephony according to terminal equipment and types of network.

2.1: PC-to-PC:

The calling and called parties both have computers that enable them to connect to the Internet, usually via the network of an Internet service provider (ISP). The two correspondents are able to establish voice communication. Both users have to be connected to the Internet at that time and use IP telephony software. In this the caller must know the IP address of the called party.



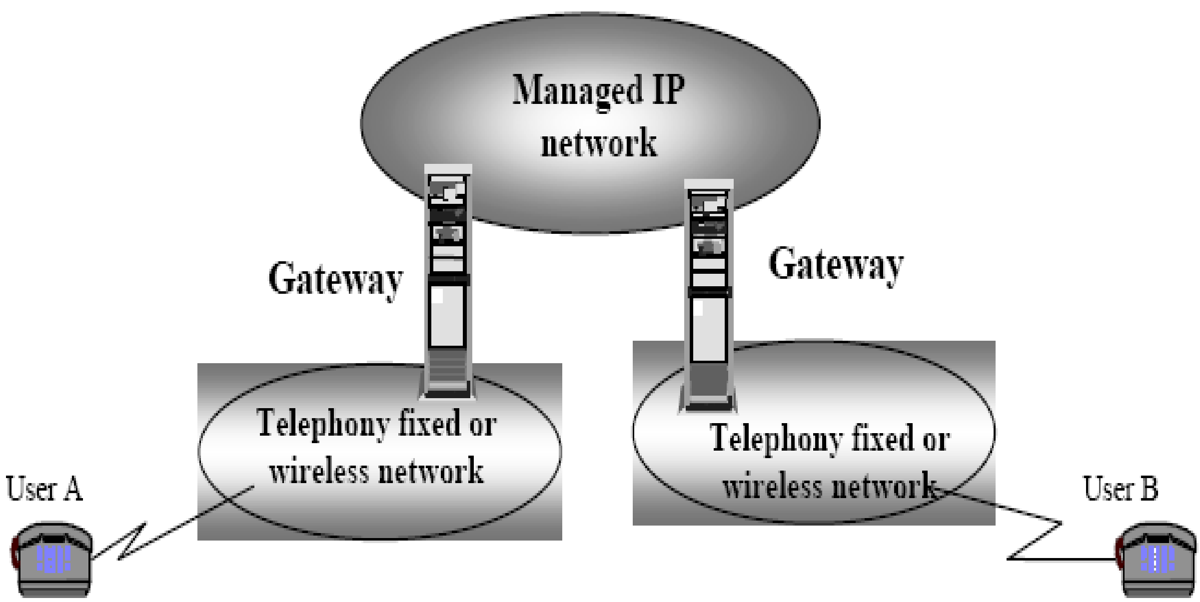
2.2. Phone-to-phone over IP:

The calling and called parties are both subscribers to the public telephony network (fixed or mobile) and use their telephone set for voice communication in the normal way.

There are two methods for communicating by means of two ordinary telephone sets via an IP or Internet network.

Use of gateways:

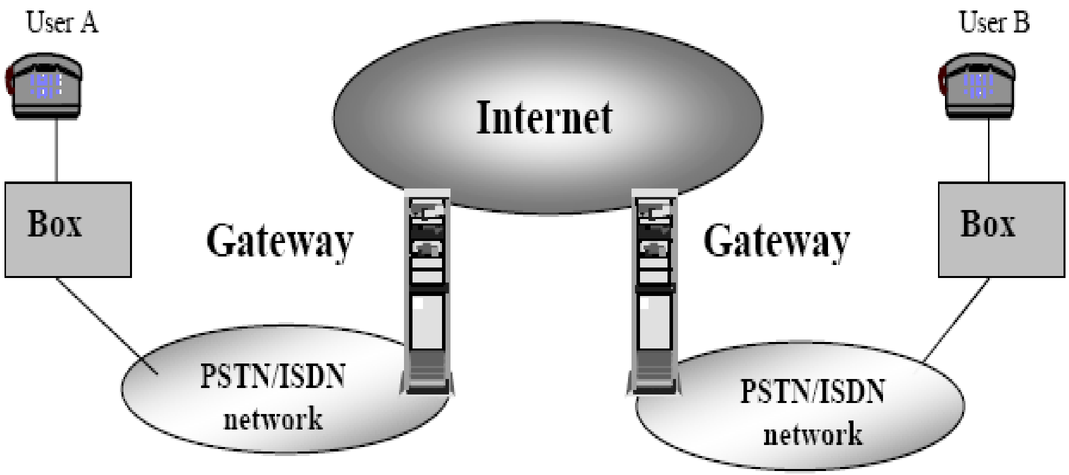
One or more telecommunication players have established gateways that enable the transmission of voice over an IP network in a way that is transparent to telephone users. It works in "managed IP network" i.e. a network, which has been dimensioned in such a way as to enable voice to be carried with an acceptable quality of service.



Use of adapter boxes:

A number of companies market boxes, which resemble modems and are installed between the user's telephone set and his connection to the PSTN.

The calling party initiates his call in the same way as in a conventional telecommunication network. The first phase of the call is set-up on that network, however, immediately after this the boxes exchange the information required for the second phase. Data they have exchanged and the pre-established parameters, establish a connection between each of the two correspondents and their respective ISP. Once the call has been established, the boxes locally convert the voice signals into IP packets to be transported over the Internet

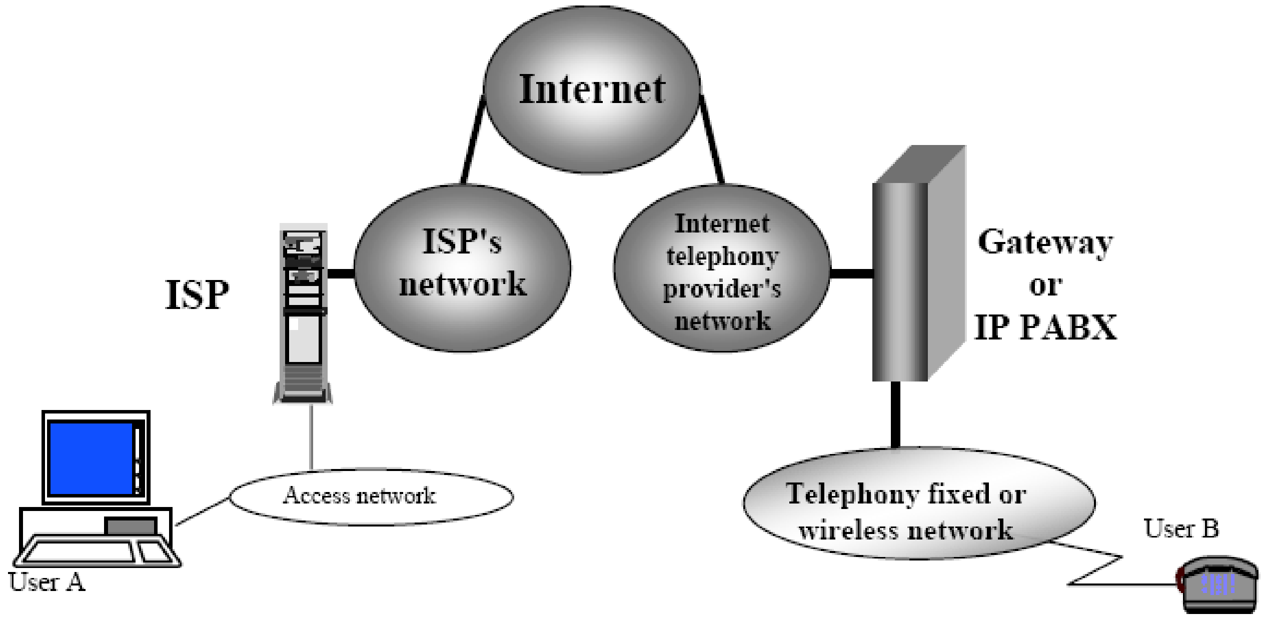


2.3. PC-to-Phone:

When the computerized user wishes to call a correspondent on the latter's telephone set, he must begin by connecting to the Internet in the traditional manner via the network of his ISP. Once connected, he uses the services of an Internet telephony service provider (ITSP) operating a gateway, which ensures access to the point that, is closest to the telephone exchange of the called subscriber. It is this gateway that will handle the calling party's call and all of the signaling relating to the telephone call at the called party end.

2.4. Phone-to-PC:

The calling party is the telephony user and the called party is the PC user.



3. Different type of standard:

Different type of standard and protocols are employed by the IP telephony. H.323, session initiation protocol (SIP), media gateway to media controller protocol(MGCP) & many more other protocol are based on IP telephony.

4. What is H.323?:

H.323 provides multimedia communication services-real-time audio, video, and data Communications-over packet networks, including Internet protocol (1P)-based networks. It specifies the components, protocols, and procedures providing multimedia communication over packet-based networks

H.323 is a standard produced by the ITU-T Study Group 16.H.323 is part of a family of ITU-T recommendations called H.32x that provides multimedia communication services over a variety of networks. H.323 can also be applied to multipoint-multimedia communications. Currently the most widely supported IP telephony signaling protocol.

5. The ITU-T H.32x Family:

The H.323 standard is part of the H.32x family of recommendations specified by ITU-T. The other recommendations of the family specify multimedia communication services over different networks:

(a)H.324 over SCN

(b)H.320 over integrated services digital networks (ISDN)

(c)H.321 and H.310 over broadband integrated services digital networks (B-ISDN)
(d)H.322 over LANs that provide guaranteed QoS
H.323 standard was interoperability with other multimedia-services networks through the use of a gateway. H.323 terminals are compatible with H.324 terminals on SCN and wireless networks, H.310 terminals on B-ISDN, H.320 terminals on ISDN, H.321 terminals on BISDN, and H.322 terminals on guaranteed QoS LANs.

6. H.323 Components:

The H.323 standard specifies four kinds of components, which, when networked together, provide the point-to-point and point-to-multipoint multimedia communication Services:
6.1. Terminals
6.2. Gateways
6.3. Gatekeepers
6.4. Multipoint control units (MCUs)
An H .323 zone is a collection of all terminals, gateways, and MCUs managed by a single gatekeeper. A zone includes at least one terminal and may include gateways or MCUs. A zone has only one gatekeeper. A zone may be independent from network topology.

6.1. Terminals:

H.323 terminal can either be a personal computer (PC) or a stand-alone device, running an H.323 and the multimedia applications. It supports audio communications and can optionally support video or data communications. Because the basic service provided by an H.323 terminal is audio communications, H.323 terminal plays a key role in IP-telephony services.

6.2. Gateways:

A gateway connects two dissimilar networks. An H.323 gateway provides connectivity between an H.323 network and a non-H.323 network. A gateway is not required, however, for communication between two terminals on an H.323 network.

6.3 Gatekeepers:

A gatekeeper can be considered the brain of the H.323 network. It has many functions

Address Translation:
The gatekeeper translates this E.164 telephone number or the alias into the network address for the destination terminal. The destination endpoint can be reached using the network address on the H.323 network.

Admission Control:
The gatekeeper can control the admission of the endpoints into the H.323 Network by using RAS messages, admission request (ARQ), confirm (ACF), and reject (ARJ).

Bandwidth Control:
The gatekeeper provides support for bandwidth control by using the RAS messages, bandwidth request (BRQ), confirm (BCF), and reject (BRJ). If a network manager has specified a threshold for the number of simultaneous connections on the H.323 network, the gatekeeper can refuse to make any more connections once the threshold is reached.

Zone Management:
The gatekeeper provides the above functions address translation, admissions control, and bandwidth control4or terminals, gateways, and MCUs located within its zone of control.

Call-Control Signaling
The gatekeeper can route call-signaling messages between H.323 endpoints using H.225 call signaling message.

Call Authorization
Gatekeeper authorizes the user to setup connection within its zone.

Call Management
The gatekeeper may maintain information about all active H.323 calls. It can control its zone by providing the maintained information.

6.4 Multipoint Control Units:
MCUs provide support for conferences of three or more H.323 terminals. All terminals participating in the conference establish a connection with the MCU. The gatekeepers, gateways, and MCUs are logically separate components of the H.323 standard but can be implemented as a single physical device.

7. Protocols Specified by H.323:
The protocols specified by H.323 are listed below.

7.1 audio CODECs

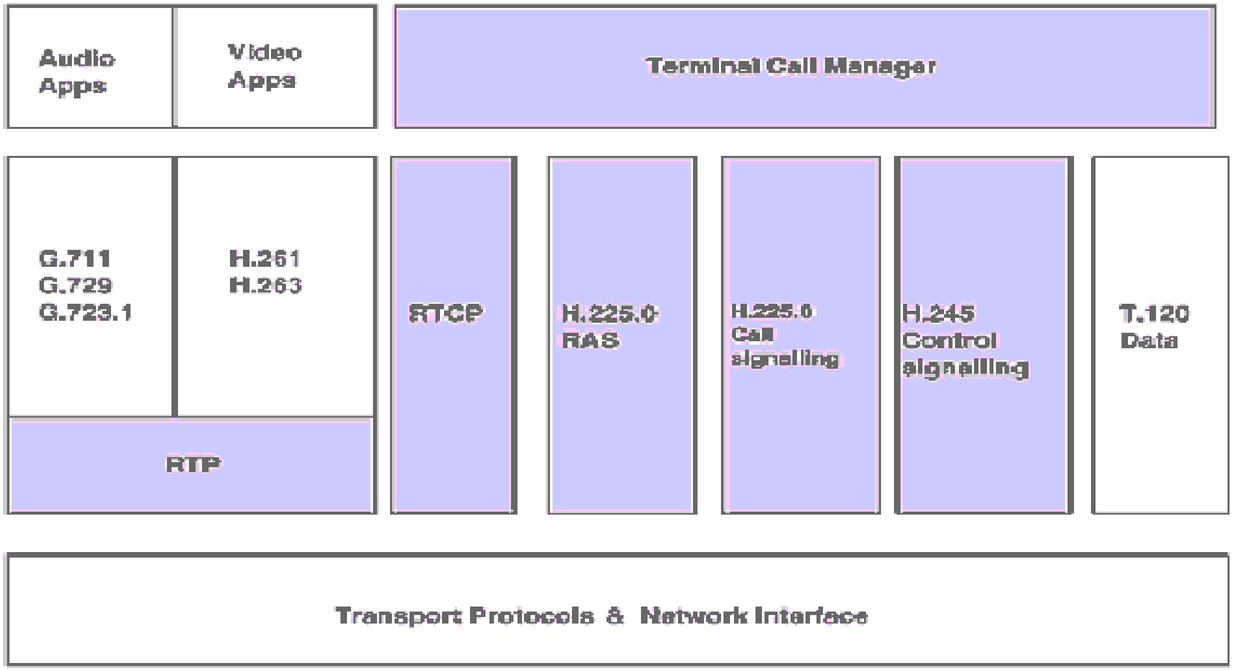
7.2 video CODECs

7.3 H.225 registration, admission, and status (RAS)

7.4 H .225 call signaling

7.5 H.245 control signaling

7.6 real-time transfer protocol (RTP)
7.7 real-time control protocol (RTCP)



7.1 Audio CODEC

An audio CODEC encodes the audio signal from the microphone for transmission on the transmitting H.323 terminal and decodes the received audio code that is sent to the speaker on the receiving H.323 terminal. Audio is the minimum service provided by the H.323 standard, all H.323 terminals must have at least one audio CODEC support. ITU-T G.711 (audio coding at 64 kbps), G.722 (64, 56, and 48 kbps), G.723.1 (5.3 and 6.3 kbps), G.728 (16 kbps), and G.729 (8 kbps) recommendation are the audio CODEC.

7.2 Video CODEC

A video CODEC encodes video from the camera for transmission on the transmitting H.323 terminal and decodes the received video code that is sent to the video display on the receiving H.323 terminal. The support of video CODECs is optional. ITU-T H.261 is the video CODEC recommendation.

7.3 H.225 Registrations, Admission, and Status (RAS)

RAS is the protocol between endpoints (terminals and gateways) and gatekeepers. RAS is used to perform these tasks

- Gatekeeper discovery (GRQ):
- Endpoint registration
- Endpoint location
- Admission control

Gatekeeper Discovery

The gatekeeper discovery process is used by the H.323 endpoints to determine the gatekeeper with which the endpoint must register.

Endpoint Registration

Registration is a process used by the endpoints to join a zone and inform the gatekeeper of the zone's transport and alias addresses.

Endpoint Location

Endpoint location is a process by which the transport address of an endpoint is determined and given its alias name or E.164 address.

Admission Control

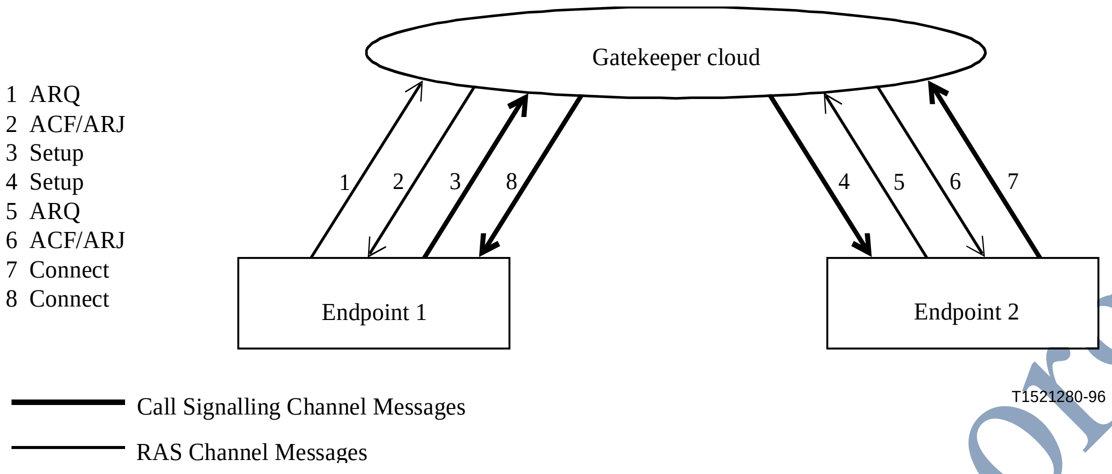
The gatekeeper can control the admission of the endpoints into the H.323 network. It uses RAS messages, admission request (ARQ), confirm (ACF), and reject (ARJ)

7.4 H.225 Call Signaling:

The H.225 call signaling is used to establish a connection between two H.323 endpoints over which the real-time data can be transported. There are the two type of Call Signaling.

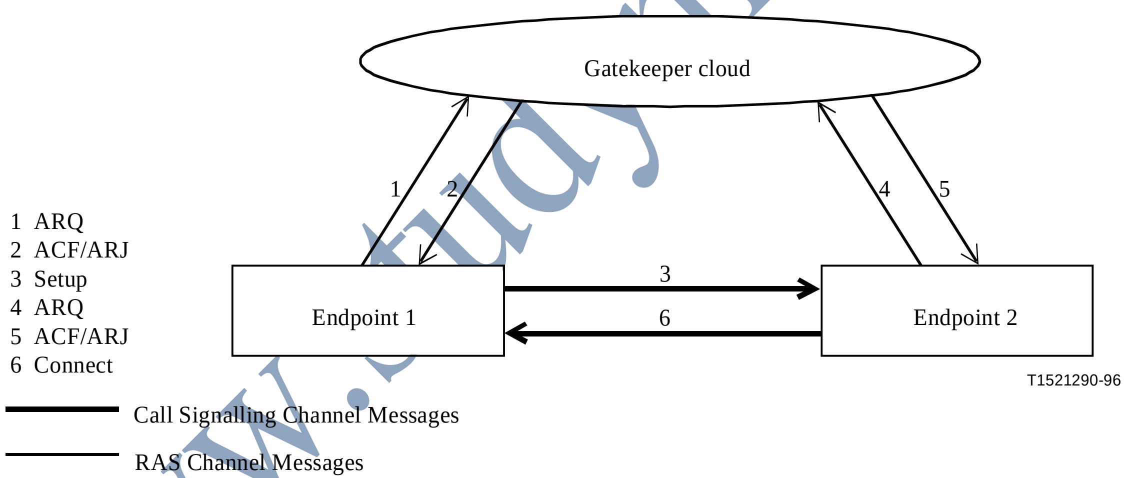
7.4.1 Gatekeeper-Routed Call Signaling

The gatekeeper receives the call-signaling messages on the call signaling channel from one endpoint and routes them to the other endpoint on the call-signaling channel of the other endpoint.



7.4.2 Direct Call Signaling

During the admission confirmation, the gatekeeper indicates that the endpoints can exchange call-signaling messages directly.



7.5 H.245 Control Signaling

H.245 control signaling consists of the exchange of end-to-end H.245 messages between communicating H.323 endpoints. The H.245 control channel is the logical channel 0 and is permanently open.

7.5.1 Capabilities Exchange

Capabilities exchange is a process using the communicating terminals' exchange messages to provide their transmit and receive capabilities to the peer endpoint.

7.5.2 Logical Channel Signaling

A logical channel carries information from one endpoint to another endpoint (in the case of a point-to-point conference) or multiple endpoints.

7.6 Real-Time Transport Protocol

Real-time transport protocol (RTP) provides end-to-end delivery services of real time audio and video. RTP, together with UDP, provides transport-protocol functionality.H.323 is used to transport data over IP-based networks; RTP is typically used to transport data via the user datagram protocol

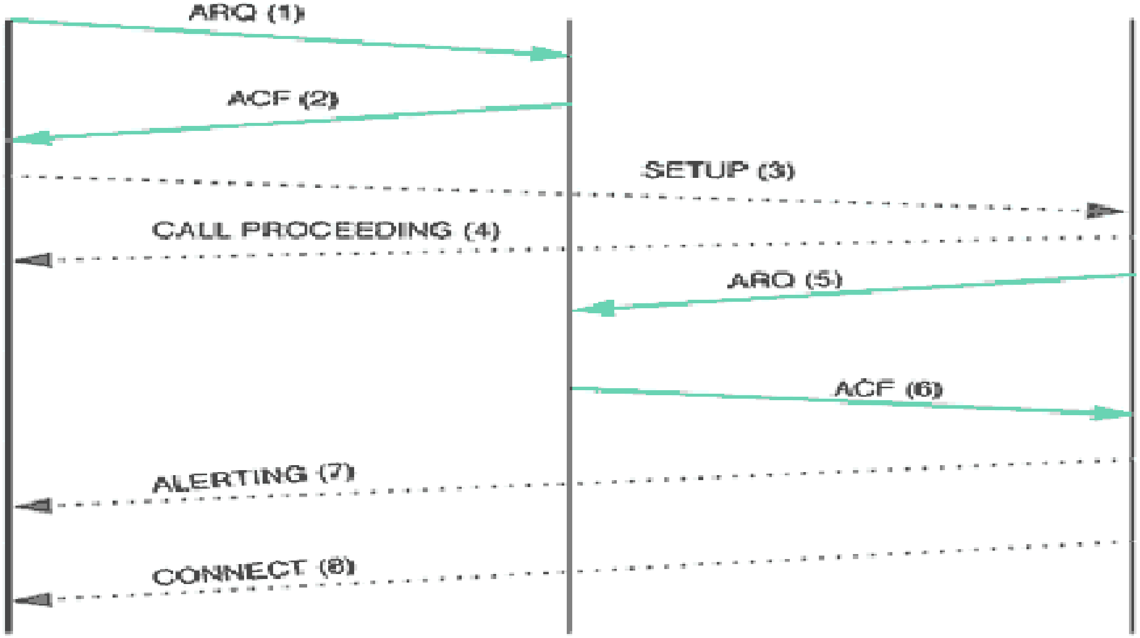
7.7Real-Time Transport Control Protocol

Real-time transport control protocol (RTCP) is the counterpart of RTP that provides control services. The primary function of RTCP is to provide feedback on the quality of the data distribution.

8. Connection Procedures:

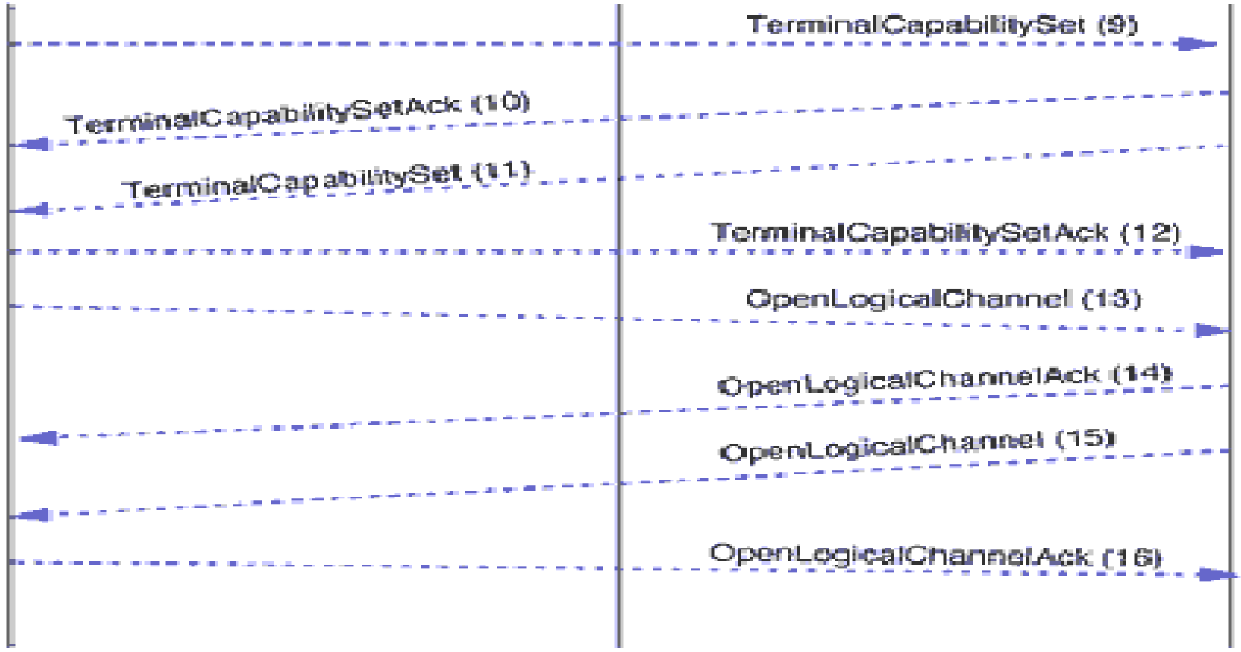
This module describes the steps involved in creating an H.323 call, establishing media communication, and releasing the call. The example network contains two H.323 terminals (TI and T2) connected to a gatekeeper. Direct call signaling is assumed. It is also assumed that the media stream uses RTP encapsulation.

Figure illustrates H.323 call establishment.



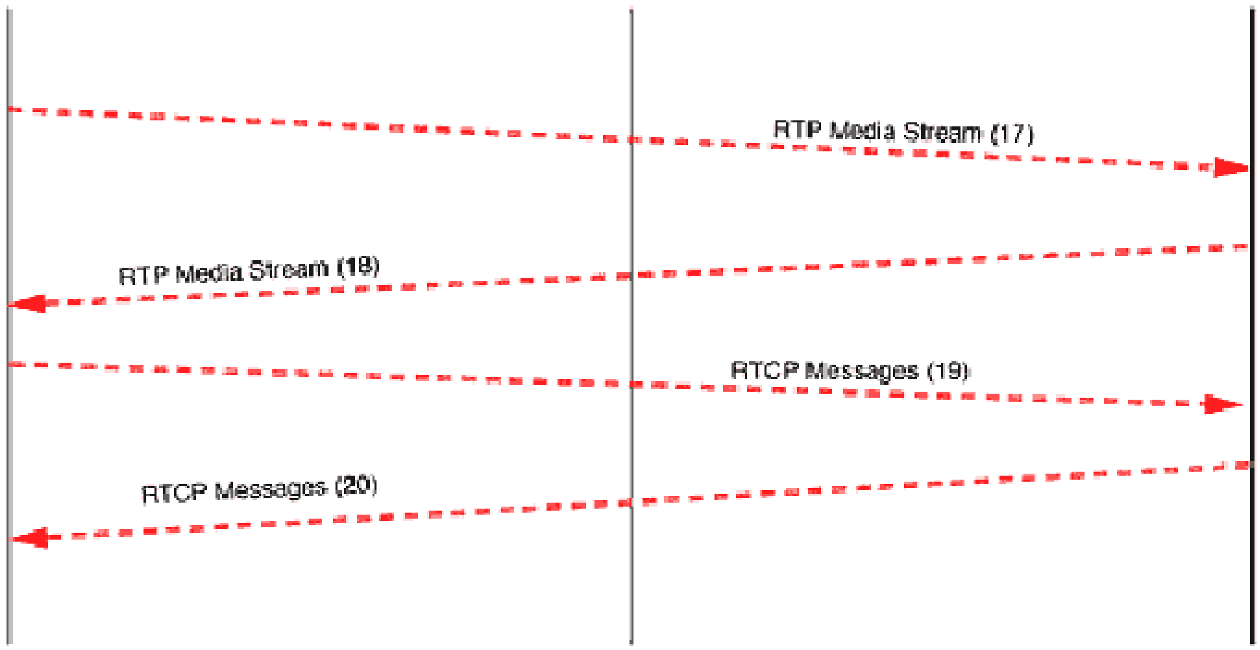
I. T1 sends the RAS **ARQ** message on the RAS channel to the gatekeeper for registration. T1 requests the use of direct call signaling.
2. The gatekeeper confirms the admission of Tl by sending ACF to T1. The gatekeeper indicates in ACF that T1 can use direct call signaling.
3. T1 sends an H.225 call signaling setup message to T2 requesting a connection.
4. T2 responds with an H .225 call proceeding message to T1.
5. Now T2 has to register with the gatekeeper. It sends an RAS ARQ message to the gatekeeper on the RAS channel.
6. The gatekeeper confirms the registration by sending an RAS ACF message to back.
7. T2 alerts T1 of the connection establishment by sending an H.225 alerting message.
8. Then T2 confirms the connection establishment by sending an H.225 connect message to TI, and the call is established.

Figure illustrates H.323 control signaling flows.



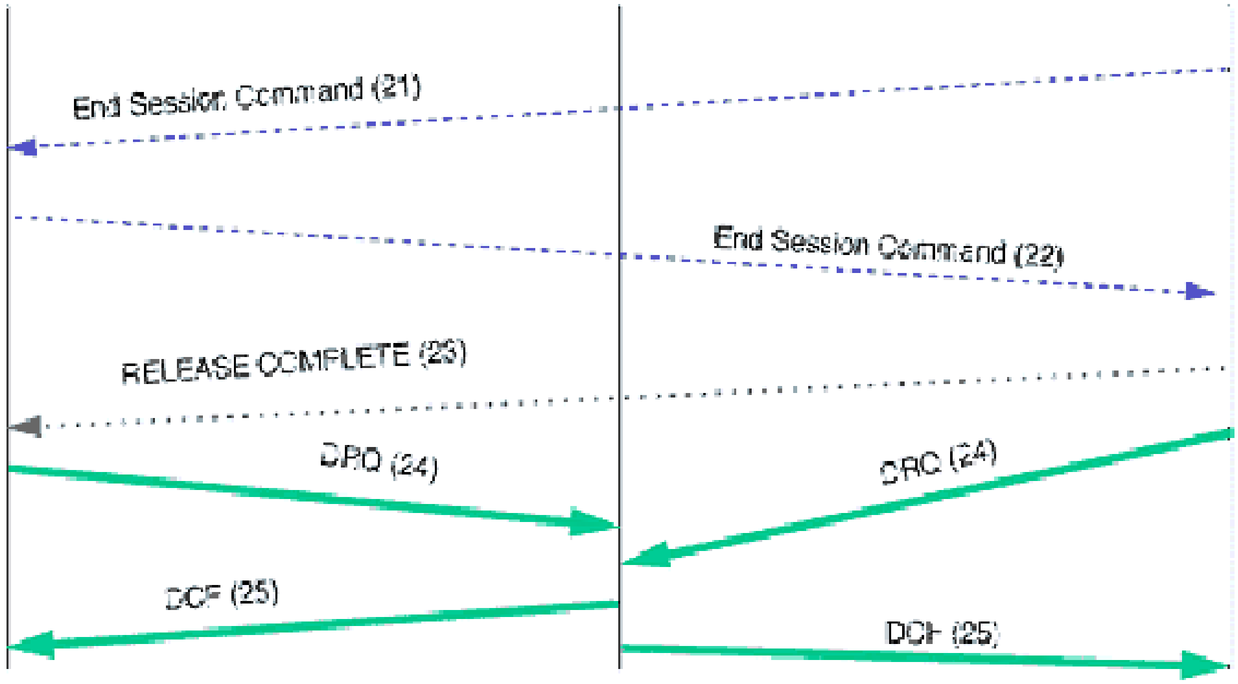
9. The H.245 control channel is established between T1 and T2. T1 sends an H.245 TerminalCapabilitySet message to T2 to exchange its capabilities.
10. T2 acknowledges Tl's capabilities by sending an H .245 TerminalCapabilitySetAck message.
11. T2 exchanges its capabilities with T1 by sending an H.245 TerminalCapabilitySet message.
12. T1 acknowledges T2's capabilities by sending an H.245 TerminalCapabilitySetAck message.
13. T1 opens a media channel with T2 by sending an H.245 openLogicalChanne1 message. The transport address of the RTCP channel is included in the message.
14. T2 acknowledges the establishment of the unidirectional logical channel from Tl to T2 by sending an H.245 openLogicalChannelAck message. Included in the acknowledge message are the RTP transport address allocated by T2 to be used by the T1 for sending the RTP media stream and the RTCP address received from T1 earlier.
15. Then, T2 opens a media channel with T1 by sending an H.245 openLogicalChanne1 message. The transport address of the RTCP channel is included in the message.
16. T1 acknowledges the establishment of the unidirectional logical channel from T2 to T1 by sending an H.245 openLogicalChannelAck message. Included in the acknowledging message are the RTP transport address allocated by TI to be used by the T2 for sending the RTP media stream and the RTCP address received from T2 earlier. Now the bidirectional media stream communication is established.

Figure illustrates H.323 media stream and media control flows.



17. T1 sends the RTP encapsulated media stream to T2.
18. T2 sends the RTP encapsulated media stream to TI.
19. T1 sends the RTCP messages to T2.
20 .T2 sends the RTCP messages to TI.

Figure illustrates call release flows.



21. T2 initiates the call release. It sends an H.245 EndSessionCommand message to TI.
22.T1 releases the call endpoint and confirms the release by sending an H.245 EndSessionCommand message to T2.
23.T2 completes the call release by sending an H.225 release complete message to TI.
24.T1 and T2 disengage with the gatekeeper by sending an RAS DRQ message to the gatekeeper.
25. The gatekeeper disengages Tl and T2 and confirms by sending DCF messages to Tl and T2.

9. Conclusion:

The Internet and IP-based networks are increasingly being used as alternatives to the public switched telephone network. IP calls can be made to almost any telephone in the world. Many public telecommunication operators are establishing their own IP telephony services, and using IP-based networks as alternative transmission platforms. There is a lot of benefits by the IP telephony. Lower infrastructure & Support costs, increase productivity, reduced cabling cost are the major benefits of IP telephony. Internet Telephony is a powerful and economical communication options by combination of the telephone networks and data networks.

10. References:

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