NRENum.net Course - Section 2 - H.323 Protocol

H.323 standard:

H.323 is a set of protocols, each one with a defined role that establishes communication between two or more communication terminals to allow the transfer of audio, video and data over a packet network.

H.323 is one of the three main standards used in videoconference communications- SIP, H.323 and MGCP. But why should H.323 be used? Because it is considered very stable and reliable.

Figure 1 shows the H.323 protocol stack in three columns.

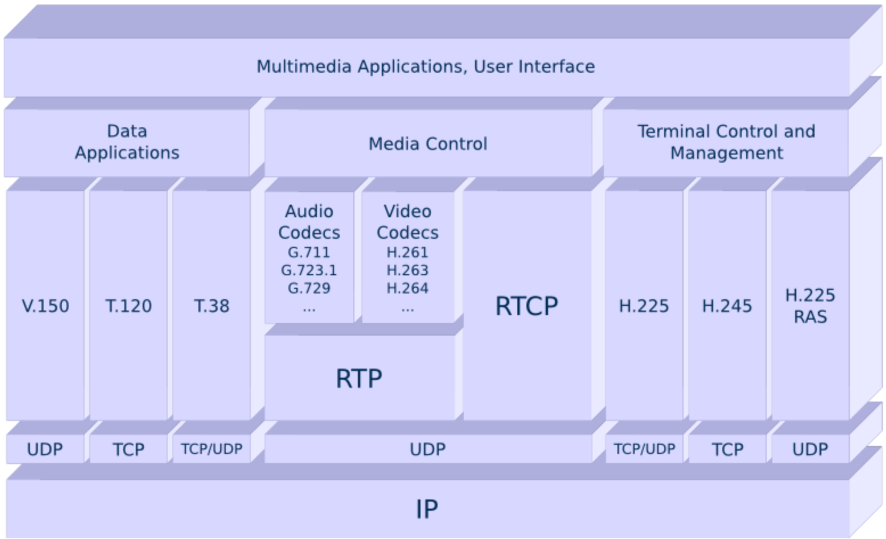

Figure 1: H.323 protocol stack

Each stack of protocols has its role to establish the communications. More information about the operation of each protocol can be found online.

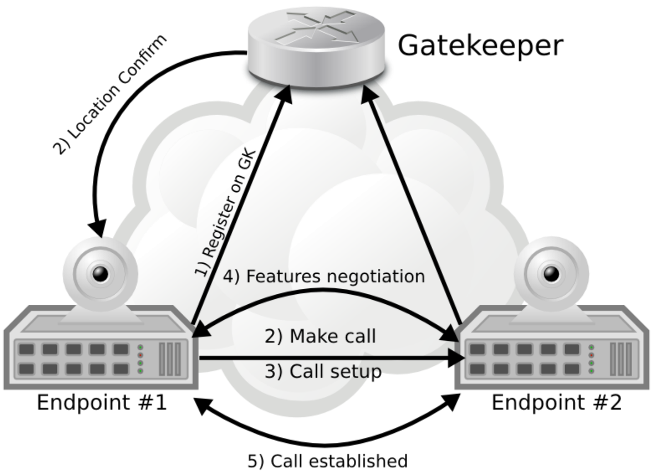

Figure 2: Process of establishing a call by using H.323 Gatekeeper

1) The first step in a call is for each terminal to be registered automatically in the Gatekeeper, which is configured previously. This process is performed by using the H.225 RAS protocol with a message called Registration Request (RR), and then the Gatekeeper agrees with Registration Confirm (RCF) or rejects with the message Registration Rejected (RRJ).

2) The second step is to dial from a terminal the number of the remote terminal to which you wish to call. This sends a message called Admission Request (ARQ) to the Gatekeeper to which the terminal is configured. If both terminals are registered in the same Gatekeeper, an Admission Confirm (ACF) is generated, otherwise, a petition called Location Request (LRQ) is generated and sent to adjacent Gatekeepers. Once the terminal is found, a Location Confirm (LCF) is answered.

3) When the terminal gets the information about remote terminal,that is when the H.225 protocol interacts. This protocol is in charge of ringing the bell at both ends, and then sets the parameters of the call (handshake negotiation).

4) At the precise moment when the call is answered, both terminals negotiate the parameters of video, audio and transmission speeds through the H.245 protocol.

5) Video and audio are transmitted during the call by using the Real-time Transfer Protocol (RTP), and monitored with a protocol called Real-time Transfer Control Protocol (RTCP).