NEXT GENERATION NETWORK

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GENERAL AGENDA

- TDM Solution
- VoIP solution H323
- SIP Solution

TDM SOLUTION

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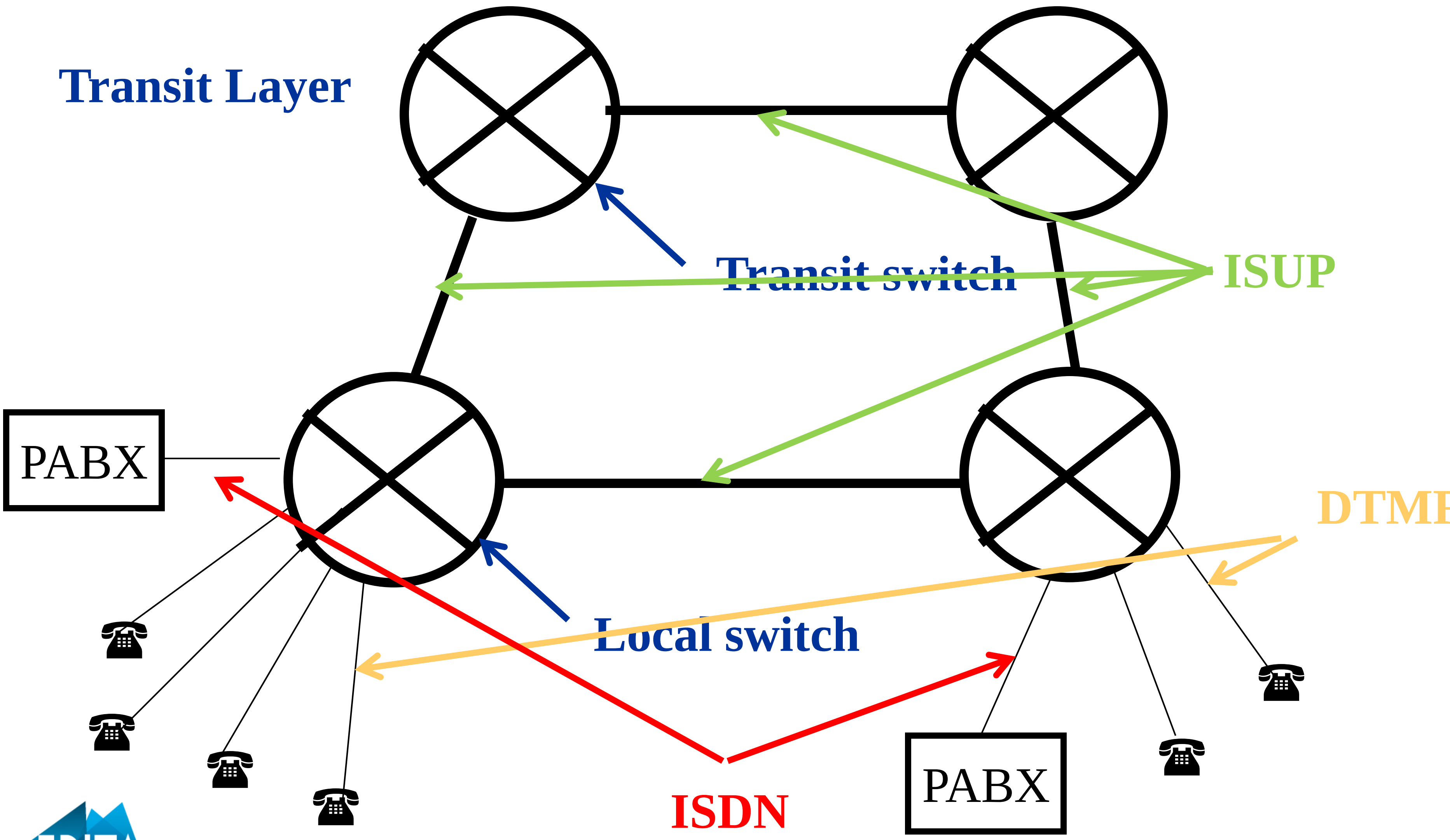
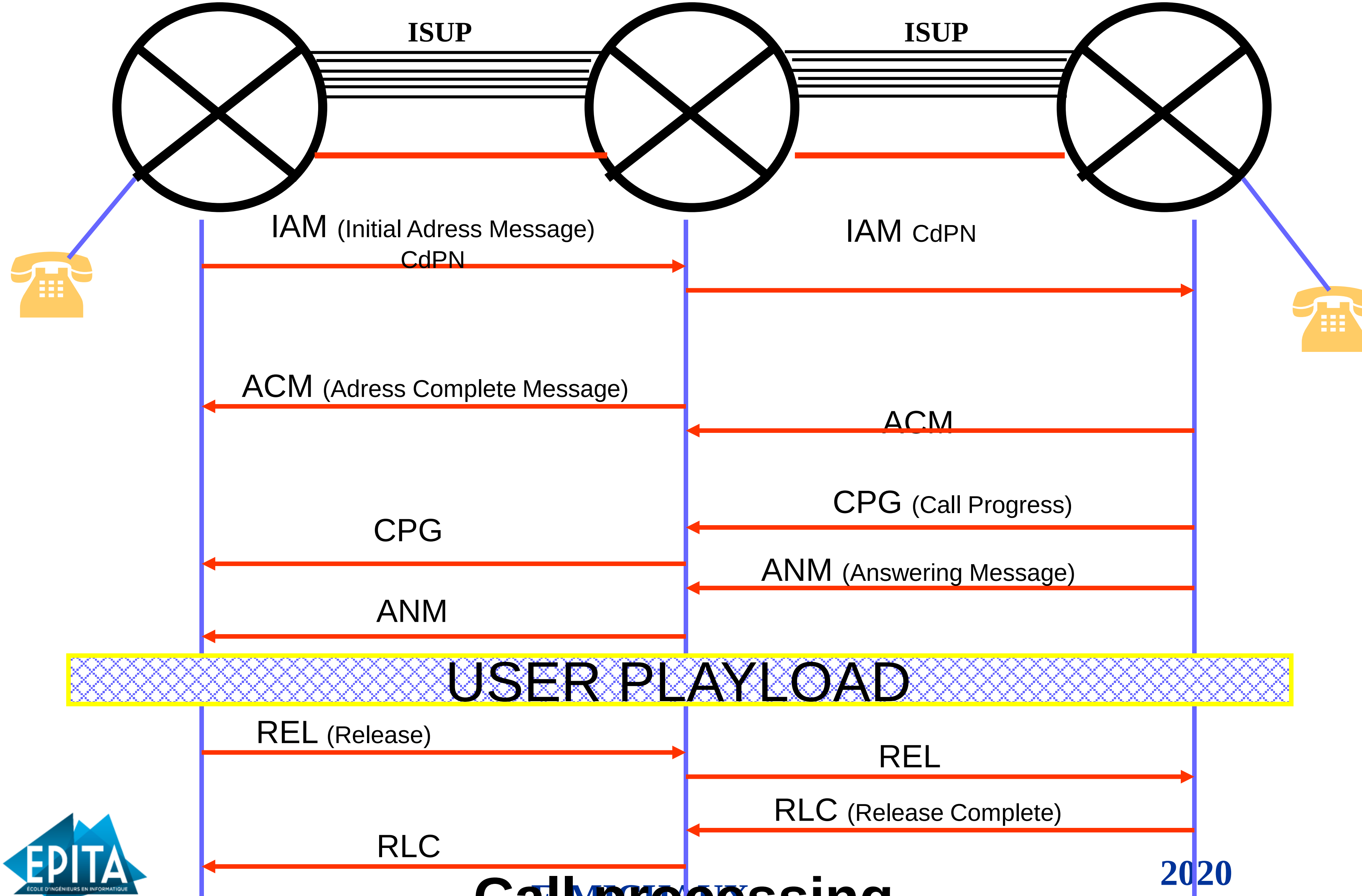
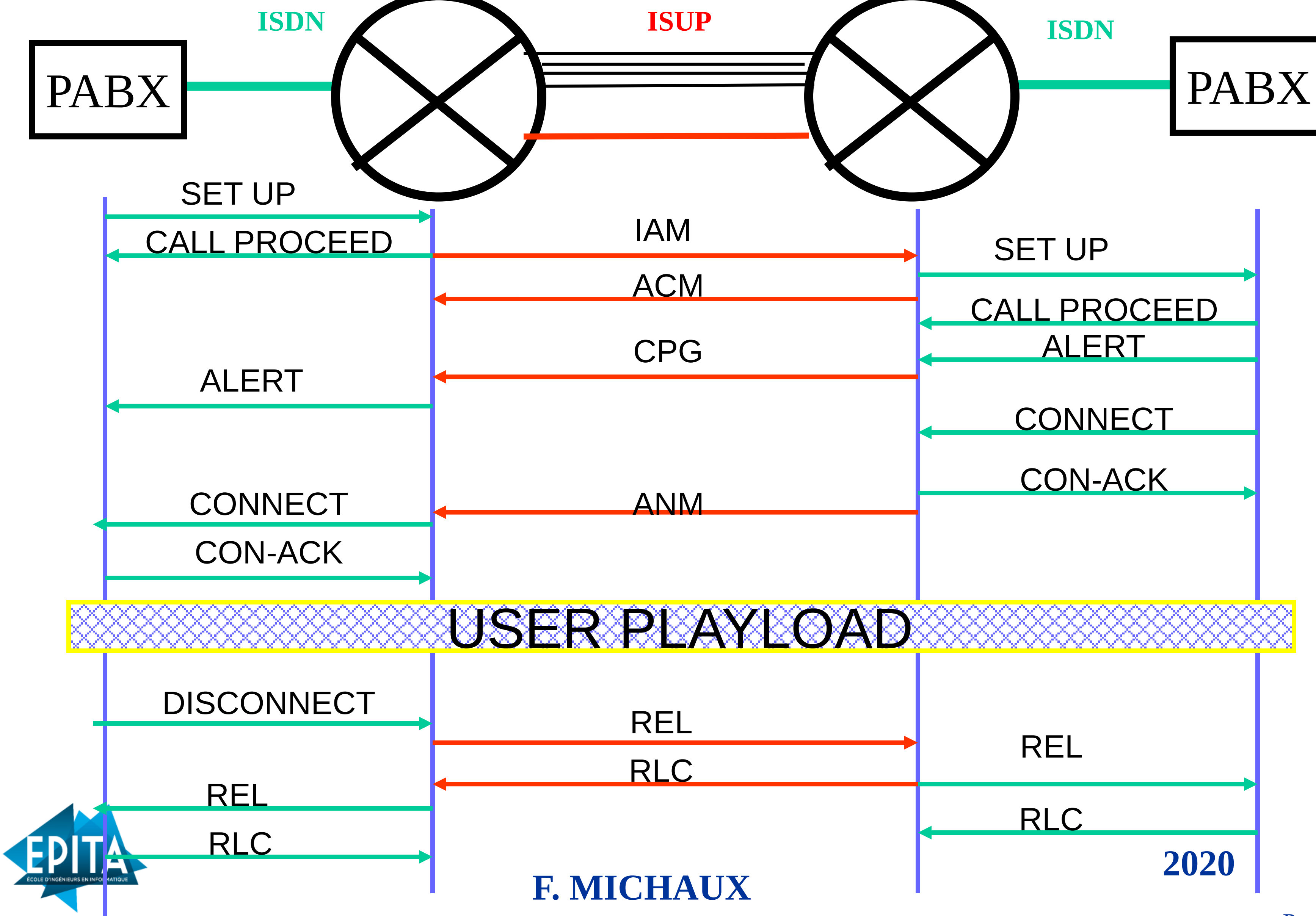
TDM Solution



ISUP Call flow



Q931 and ISUP Call Flow



VoiP SOLUTION
H323

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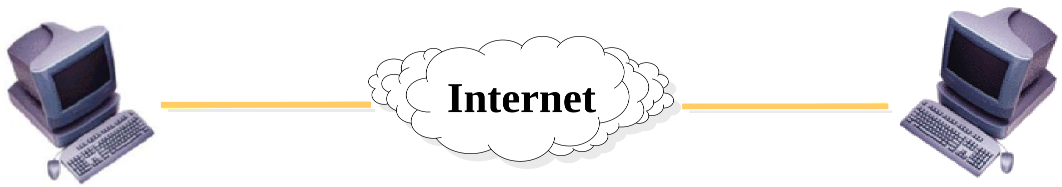
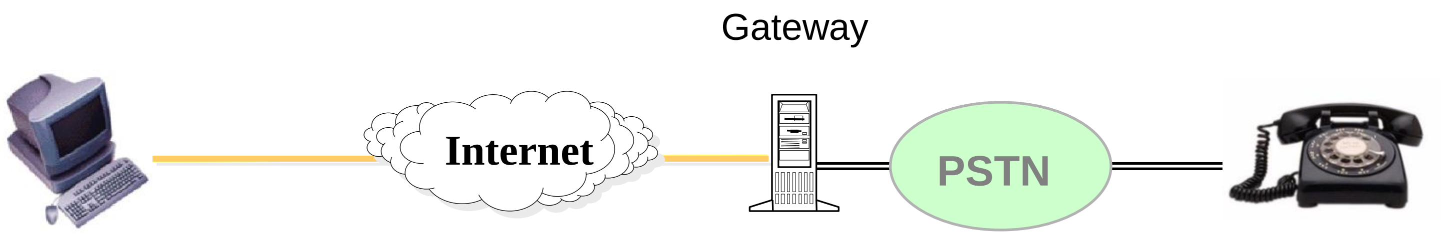
Different Type of VoIP H323 solution

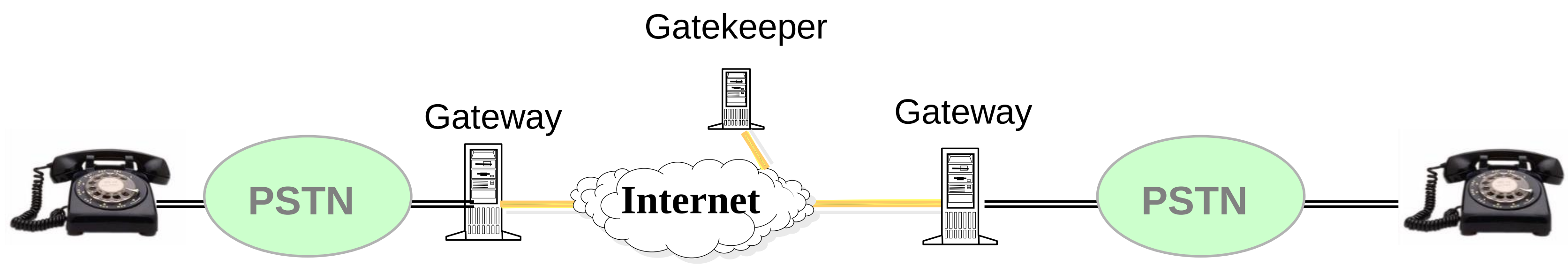


VoIP : IP Terminal to IP Terminal

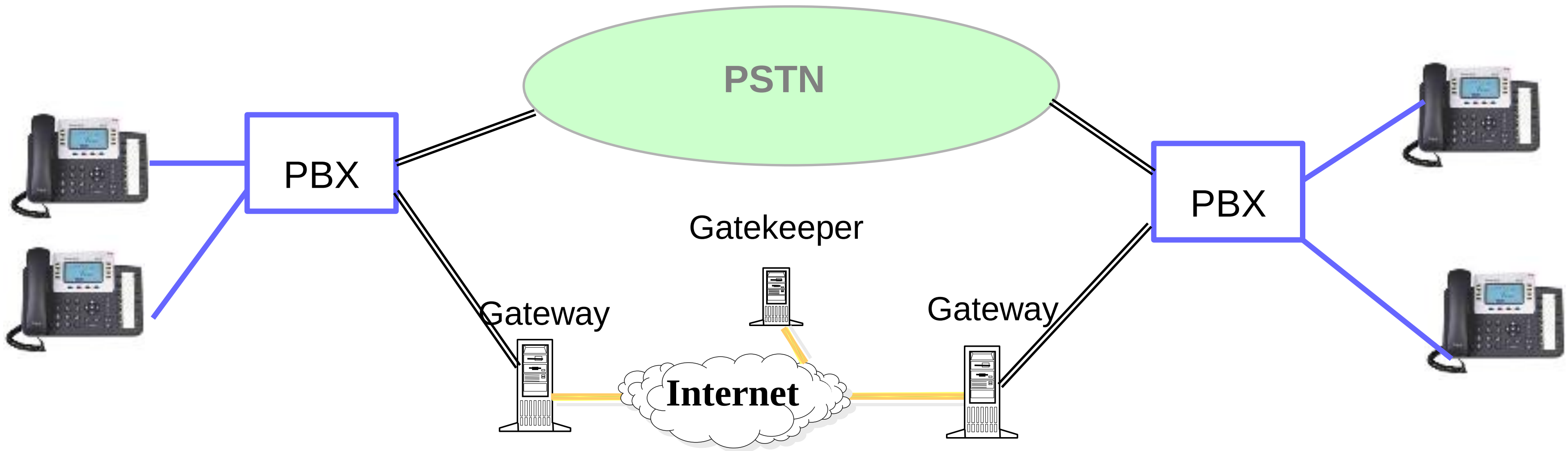


VoIP : IP Terminal to PSTN network

Different Type of VoIP H323 solution



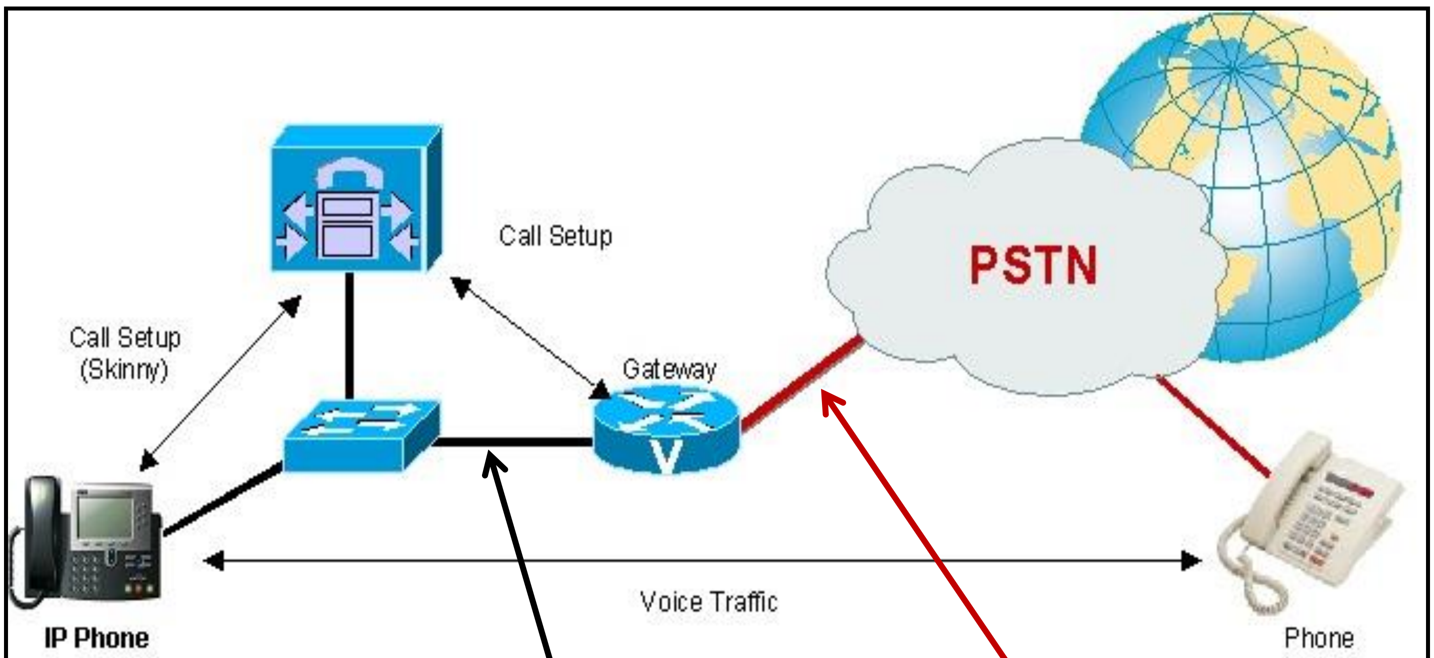
VoIP : IP Terminal to Phone via a PSTN network



VoIP in transit for Corporate PBX

VoIP Gateway

A **VoIP gateway** is a gateway device that uses Internet Protocols to transmit and receive voice communications (VoIP). The general term is ambiguous and can mean many different things. There are many such devices. They are quickly becoming the most common type of voice phone service in many areas.



UDP Port Ori/Dest Voice path

CIRCUIT ID Voice path is 2 X 64 Kbits/s

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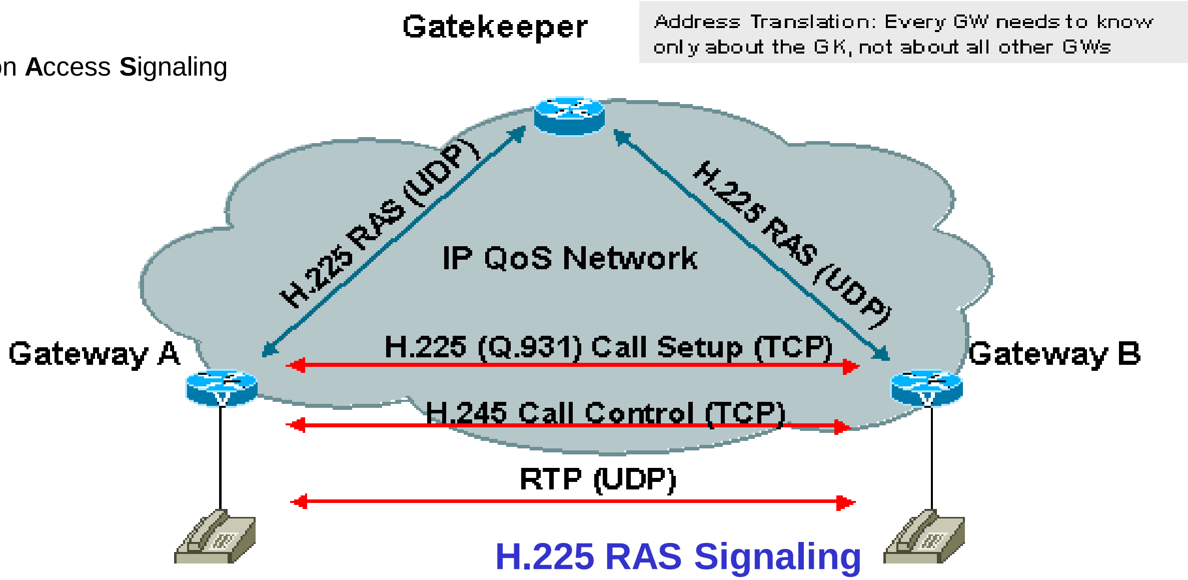
H323 Description

R.A.S : **R**egistration **A**ccess **S**ignaling



H.225 RAS Signaling

RAS is the signaling protocol used between gateways and gatekeepers. The RAS channel is opened before any other channel and is independent of the call setup and media transport channels.
RAS uses User Datagram Protocol (UDP) ports 1719 (H.225 RAS messages) and 1718 (multicast gatekeeper discovery).

H.225 Call Control (Setup) Signaling

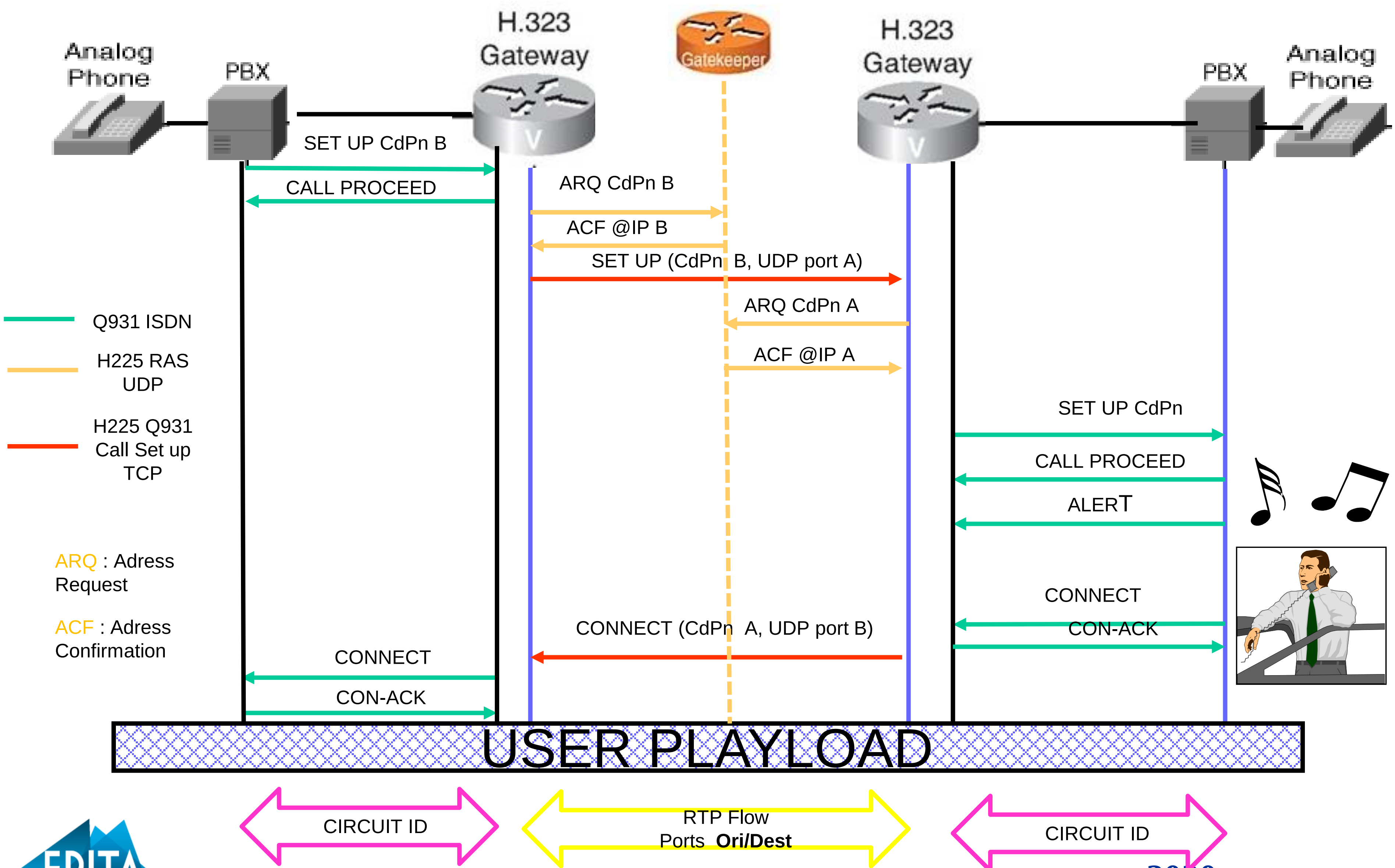
H.225 call control signaling is used to setup connections between H.323 endpoints. The ITU H.225 recommendation specifies the use and support of Q.931 signaling messages.
A reliable (TCP) call control channel is created across an IP network on TCP port 1720. This port initiates the Q.931 call control messages for the purpose of the connection, maintenance, and disconnection of calls.

H.245 Media Control and Transport

H.245 handles end-to-end control messages between H.323 entities. H.245 procedures establish logical channels for transmission of audio, video, data, and control channel information. It is used to negotiate channel usage and capabilities such as:

- flow control
- capabilities exchange messages

H323 Call flow from PBX to PBX (Faststart pro.)



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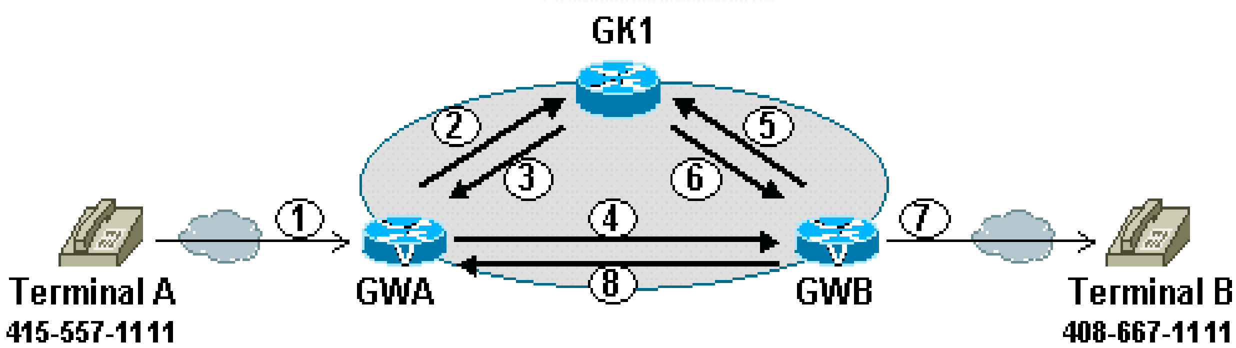
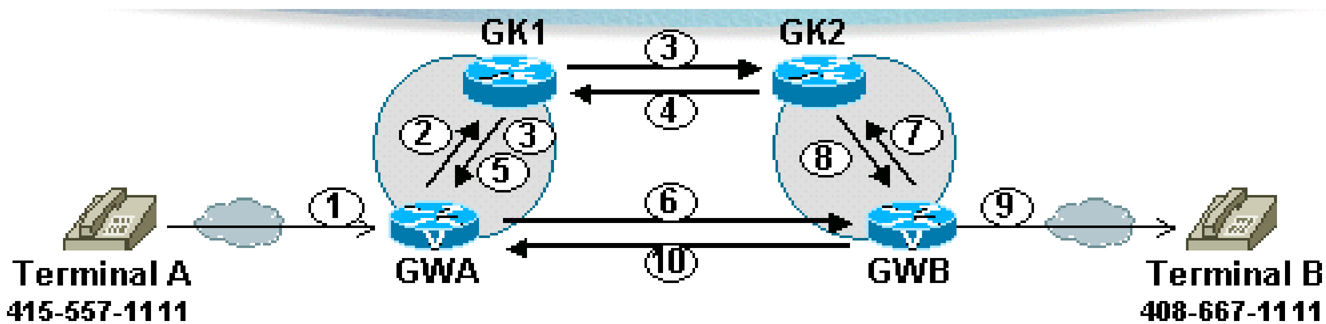
H323 Intra-zone Call Set Up



1) Terminal A dials the phone number 408-667-1111 for Terminal B
2) GWA sends GK1 an ARQ, asking permission to call Terminal B
3) GK1 does a look-up and finds Terminal B registered; returns an ACF with the IP address of GWB
4) GWA sends a Q.931 Call-Setup to GWB with Terminal B's phone number
5) GWB sends GK1 an ARQ, asking permission to answer GWA's call
6) GK1 returns an ACF with the IP address of GWA
7) GWB sets up a POTS call to Terminal B at 408-667-1111
8) When Terminal B answers, GWB sends Q.931 Connect to GWA
9) GWs sends IRR to GK after call is setup

H323 Inter-zone Call Set Up



1) Terminal A dials the phone number 408-667-1111 for Terminal B
2) GWA sends GK1 an ARQ, asking permission to call Terminal B
3) GK1 does a look-up and does NOT find Terminal B registered; GK1 does a prefix look-up and finds a match with GK2; GK1 sends an LRQ GK2, and RIP (Request In Progress) to GWA
4) GK2 does a look-up and finds Terminal B registered; returns an LCF with the IP address of GWB
5) GK1 returns an ACF with the IP address of GWB
6) GWA sends a Q.931 Call-Setup to GWB with Terminal B's phone number
7) GWB sends GK2 an ARQ, asking permission to answer GWA's call
8) GK2 returns an ACF with the IP address of GWA
9) GWB sets up a POTS call to Terminal B at 408-667-1111
10) When Terminal B answers, GWB sends Q.931 Connect to GWA

VoiP SOLUTION
SIP

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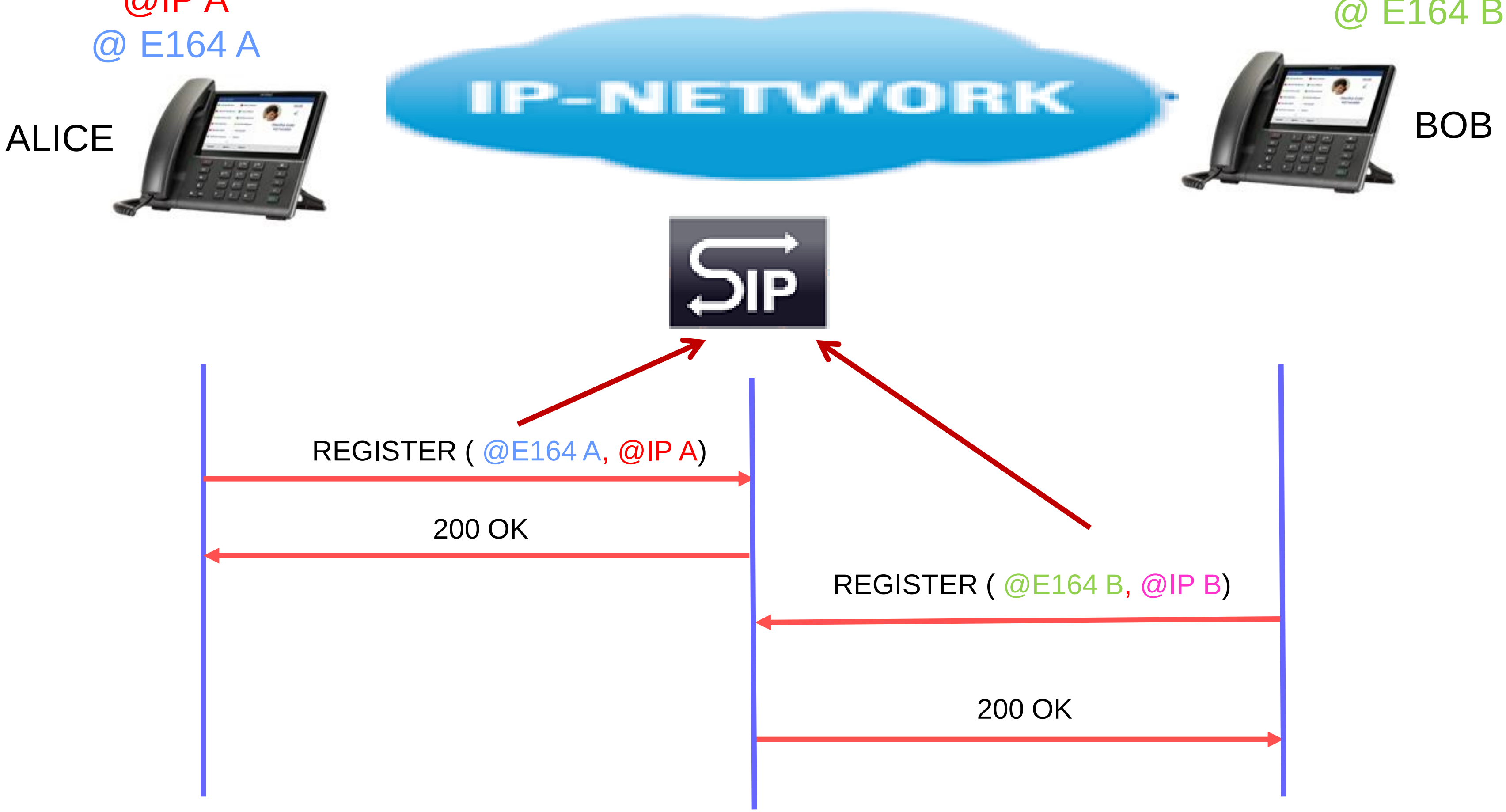
SIP Session Initiation Protocol

The Session Initiation Protocol (SIP) is a communications protocol for signaling and controlling multimedia communication sessions. The most common applications of SIP are in Internet telephony for voice and video calls, as well as instant messaging, over Internet Protocol (IP) networks.

The protocol defines the messages that are sent between endpoints, which govern establishment, termination and other essential elements of a call. SIP can be used for creating, modifying and terminating sessions consisting of one or several media streams. SIP is an application layer protocol designed to be independent of the underlying transport layer.

SIP works in conjunction with several other application layer protocols that identify and carry the session media. Media identification and negotiation is achieved with the Session Description Protocol (SDP). For the transmission of media streams (voice, video) SIP typically employs the Real-time Transport Protocol (RTP) or Secure Real-time Transport Protocol (SRTP). For secure transmissions of SIP messages, the protocol may be encrypted with Transport Layer Security (TLS).

Phone Registration SIP



Full SIP call