

## Introduction to VoIP

RFCs (RTP, SIP, H.323)

Various books on VoIP

## Telephony

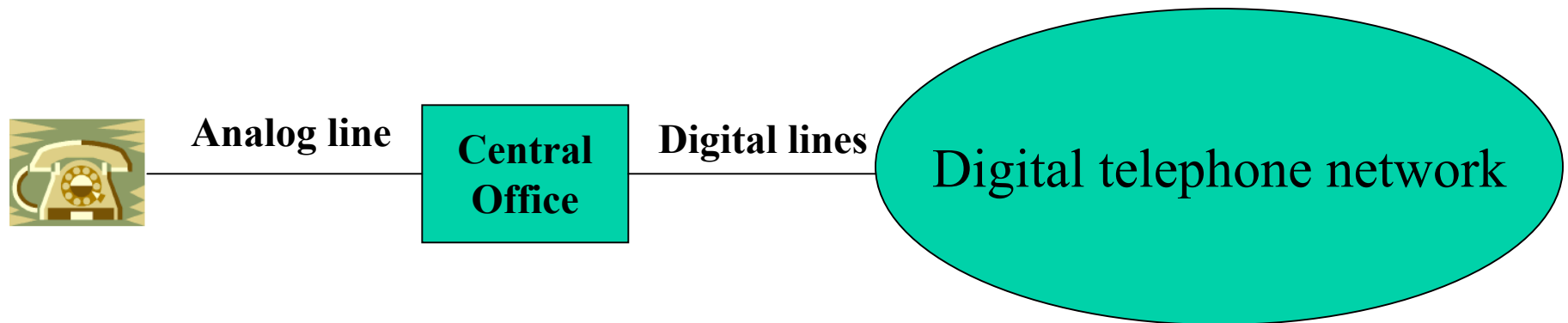


Manual operators on switchboard or  
Electromechanical gear

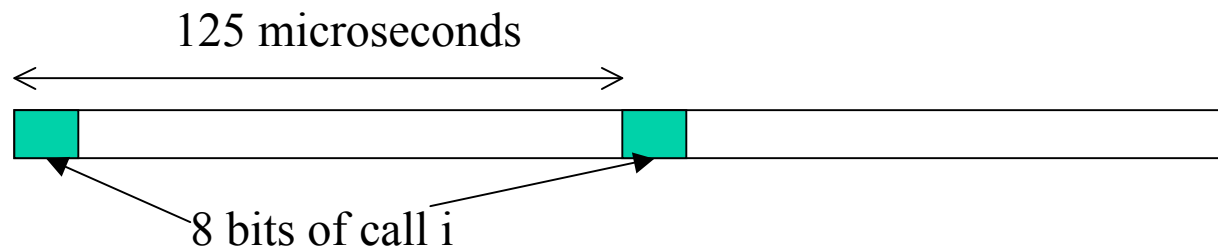
Problems: Maintenance – nightmare

Noise – multiplex, diff voice circuits on diff freq

## Digital Telephony



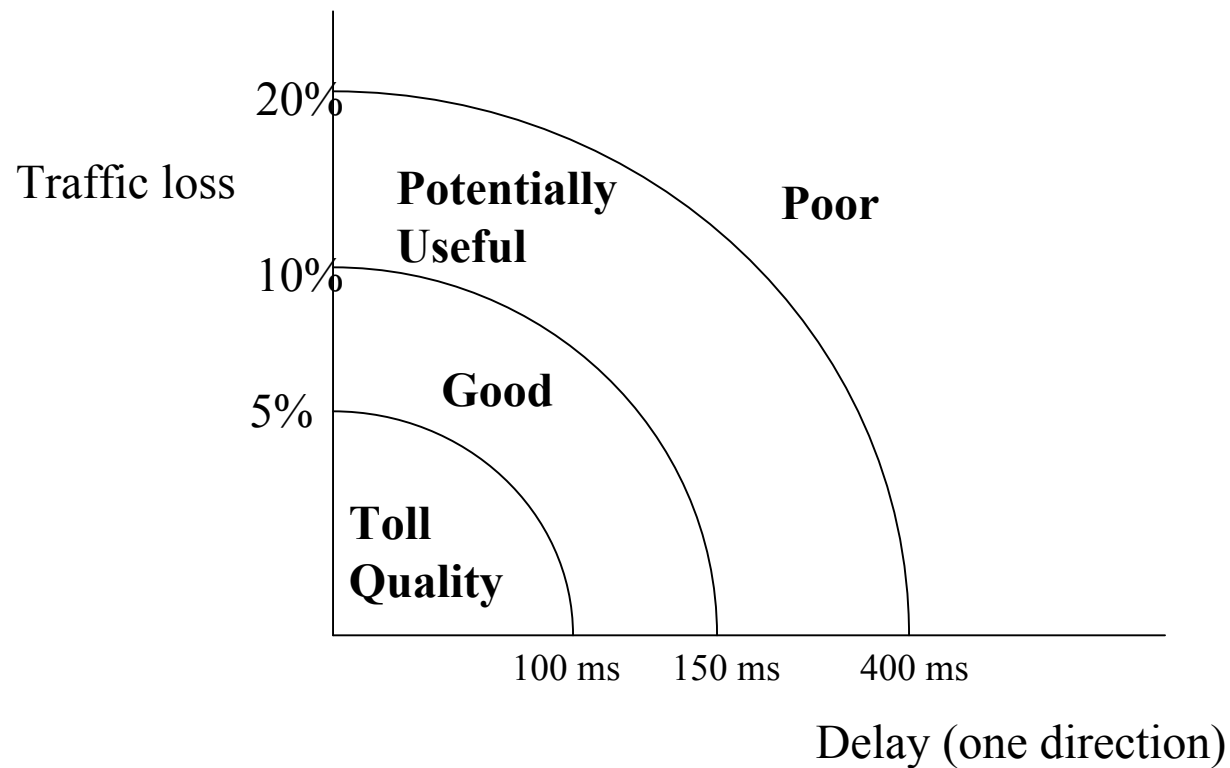
Digitize voice [8 bits/sample, 8000 samples/s]  
Multiplex voice samples



## Statistical MUX/Asynchronous Transmission

- g Approximately 35% of time we talk; other 65% of time we listen and pause
- g Why transmit silence? Transmit someone else's speech or data
- g Need less bandwidth or pack more calls
- g Uncertainty, jitter

## Call Quality



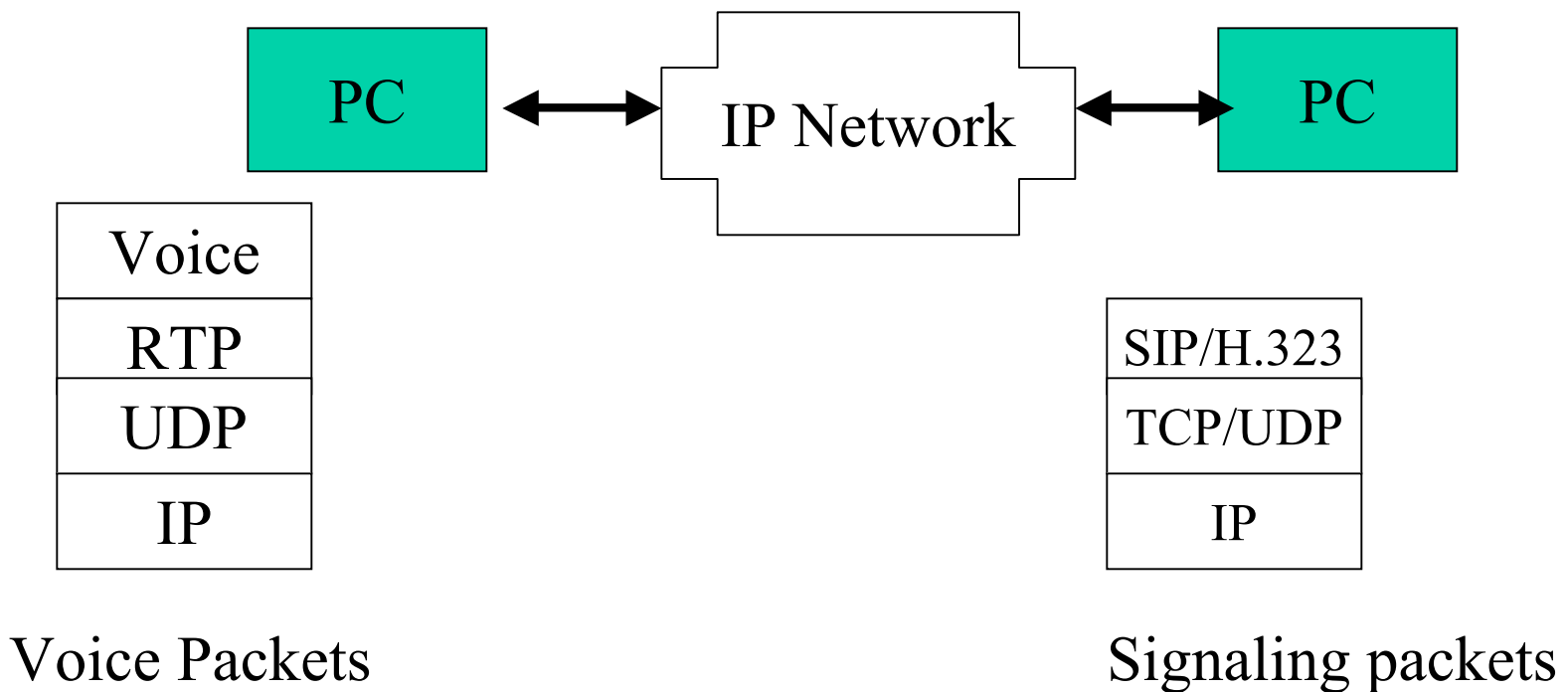
## Need for VoIP

- g Voice is digitized and transmitted in current telecom networks anyway
- g Integrate voice and data (no need to build and maintain two separate networks)
- g Make use of statistical multiplexing and use bandwidth more effectively
- g Silence, mostly half-duplex conversation, etc, can be exploited.

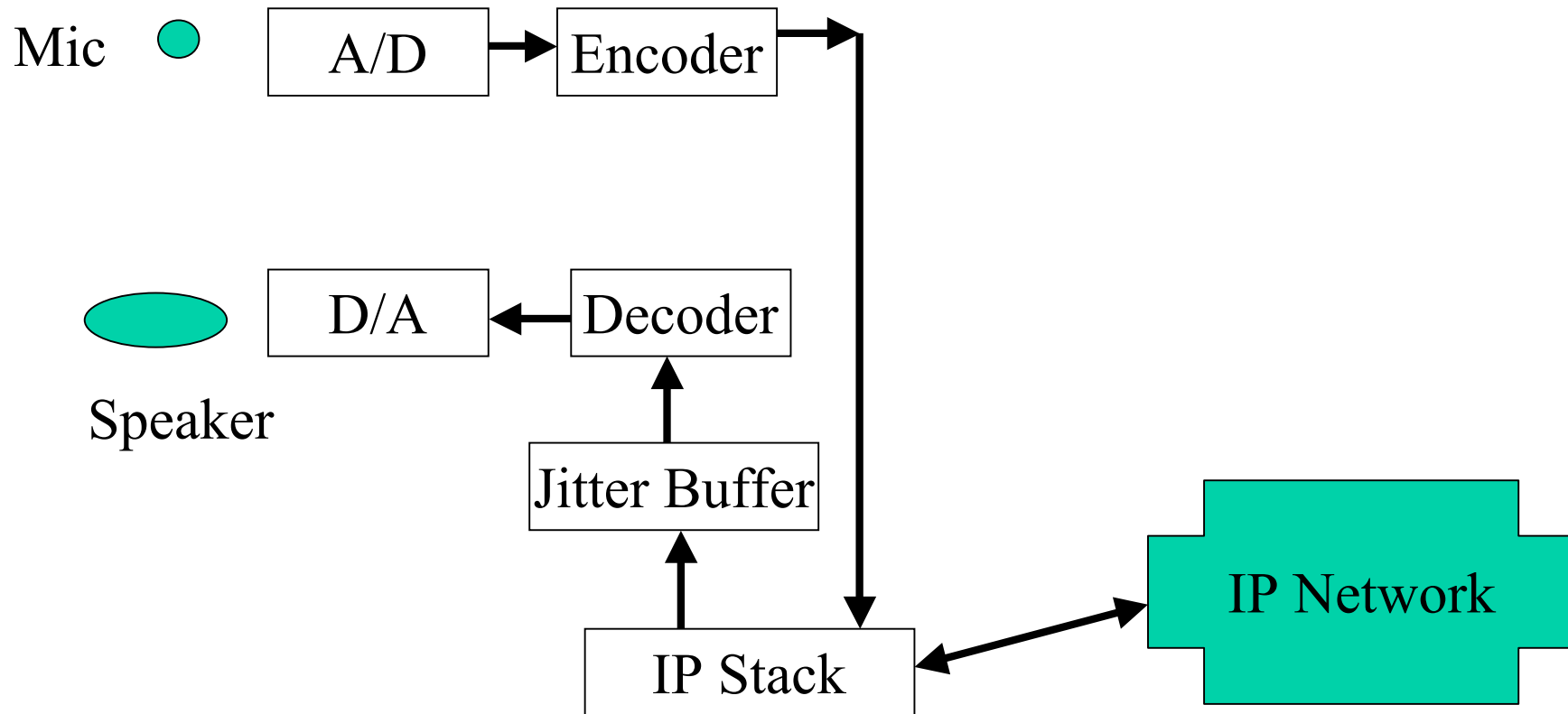
## Main issues

- g Encoding/decoding
- g voice quality
- g Loss of voice packets
- g Variable delay of the IP network (between packets of the same conversation); jitter
- g End-to-end delay (because of router delays, encoding/decoding delays), etc.
- g Data and signaling parts
- g VoIP <> PSTN (Public Switched Telephone Network)

## Protocols Used (PC-PC call)



## Architecture



## RTP (Real-Time Transport Protocol)

- g Receiver can compensate for jitter (for both voice and video)
- g RTCP (Real Time Control Protocol) is used in conjunction with RTP (convey quality of transmission, such as average packet loss, amount of jitter, etc.)
- g RTP can be multicast (destination address = multicast address)

## RTP

- g Defines a way to format IP packets carrying time-sensitive data
- g Important fields:
  - Type of data carried
  - time-stamp of sampled data (to replay correctly with correct timing between coded packets)
  - Sequence numbers (detect loss; copy last frame and repeat for gaps or interpolate for missing frames)

## Payload Type

- g Application determines contents
- g Carry payload type in RTP headers
- g Application does not need to examine contents.
- g RTP Session: Association of participants (need two transport addresses, RTP/RTCP)

## RTP Header

V (2)	P(1)	X(1)	CC(4)	M(1)	PT(7)	Sequence number (16)
Time Stamp						
Synchronization Source Identifier						
Contributing Source Identifier 1						
Contributing Source Identifier n						
Profile Dependent					Size	
Data						

## RTP Header

- g Version (2 bits); currently = 2
- g P: Padding bit: 1=payload has been padded for alignment (Last octet gives the number)
- g X: extension bit. If 1, extension is present, fixed single xtn must follow header
- g cc: Contributing SouRCe count
- g M: Mark bit: If 1, this is the first packet after silence. (no packet during silence)
- g PayloadType: can vary dynamically
  - 0=PCM mu law
  - 8=PCM A law
  - 9=G.722 ...

## RTP Header Cont'd

- g Seq #: start at random number; increment by 1 for each packet
- g time stamp[32 bits]: time value when first sample in payload was sampled;
  - initial value chosen randomly
- g SSRC: ID of source of RTP stream
- g CSRC: Contributing SouRCe (If RTP stream is a combination of several streams, ids of senders)

## RTCP Real-Time Transport Control Protocol

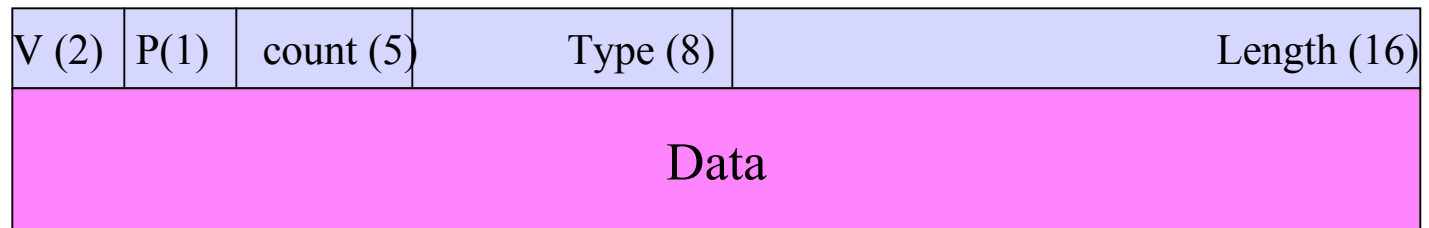
Gathers stats about quality of network;  
periodically sends to sender of RTP packets

Use: Adjust QoS parameters

Types of reports:

- g Sender Report (SR) (by active senders)
- g Receiver Report (RR) (by participants that are not active senders)
- g Source description-various parameters about SRC
- g Bye - End of participation
- g Application specific

## RTCP Header



V: Version (Same version as RTP)

P: Padding; 1 = Data is padded. (Last byte shows # of padded bytes)

Count: Number of reception blocks

Type: Type of report

Length: Length of this RTCP packet in 32 bit words minus one, including the header and any padding

## RTCP Header

- g Version
- g P (1=padding used)
- g RC (count): How many receiver blocks in packet
- g PT: Payload type (200=SR, 201=RR, ...)
- g Length: Of sender report
- g NTP Time Stamp: Network Time Protocol time stamp
- g RTP Time Stamp
- g Sender's packet count
- g Sender's octet count

## RTCP Header Cont'd

### g RR Data:

- Fraction Lost
- Total number of packets lost
- Highest sequence number received
- Interarrival jitter [average] see next
- Last SR time stamp NTP time stamp received as part of most recent RTP sender report from SSRC
- Delay since last SR: number of  $(1/65536)$  seconds delay between receiving last SR from this SSRC and sending this report

## Jitter Calculation

g  $S_i$  = Time Stamp of sender of packet  $i$

g  $S_j$  = Time Stamp of sender of packet  $j$

g  $R_i$  = Time Stamp of receiver of packet  $i$

g  $R_j$  = Time Stamp of receiver of packet  $j$

g  $D[i,j] = (R_j - R_i) - (S_j - S_i)$ : Interarrival jitter

g Cumulative Jitter  $J = J + (|D[i-1,j]| - J) / 16$

## Echo Handling

- g Traditionally, 4 wire to 2 wire transformation adds echo
  
- g Two types of echo:
  - Electric echo
  - Acoustic echo

## Acoustic Echo

- g Time between speech and the same sound getting into the mic:
  - $\leq 20$  milliseconds? No noticeable echo
  - $\geq 40$  ms? Problem
  - Speaker phone is also a source of echo

## Echo Suppressers:

- g Introduce large loss in path from me to far end if far end person is talking
- g Squelch if both talk at the same time

## Echo Cancellers

- g Build an estimate of echo (based on what goes out of the mic and the delay)
- g Remove from incoming signal at the correct time

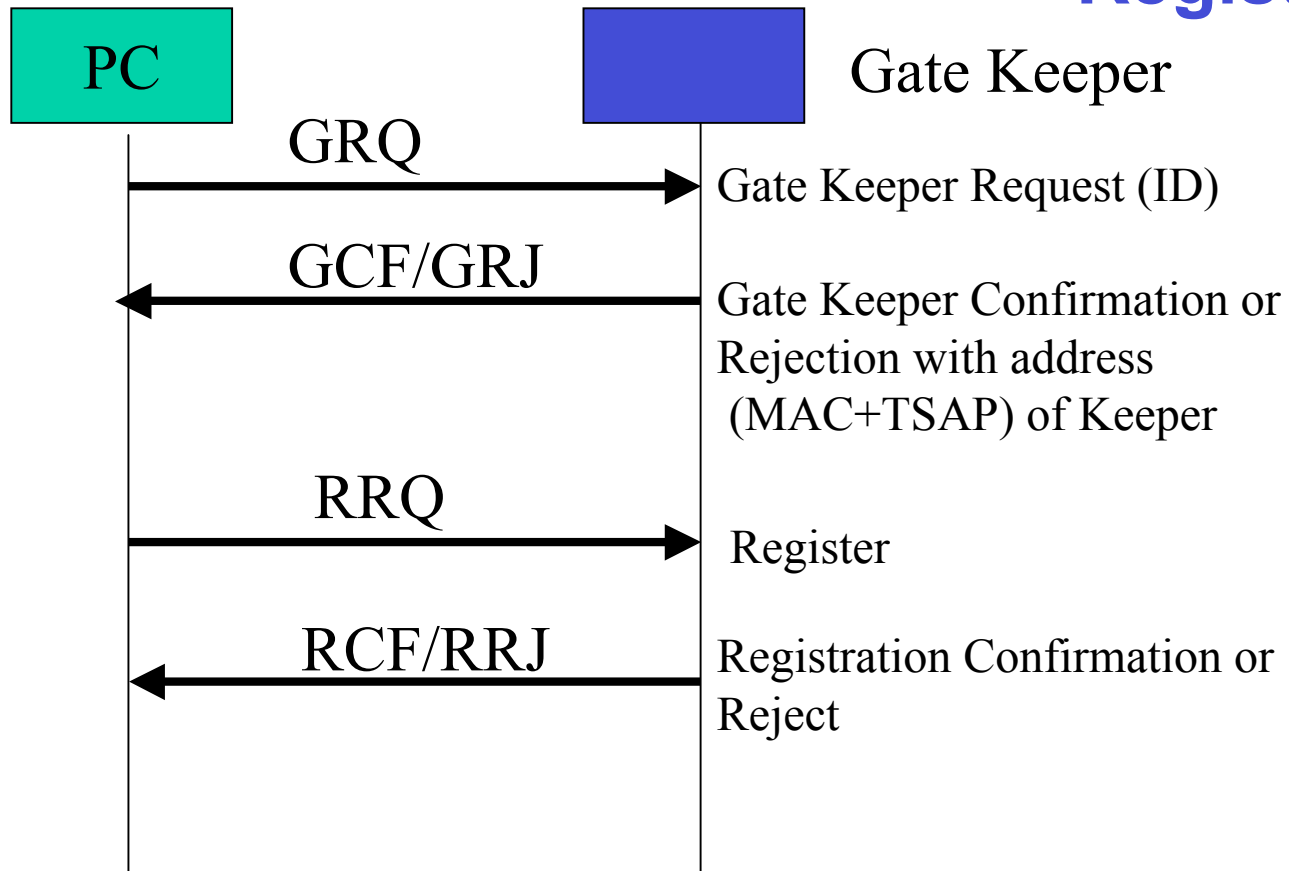
## Signaling

- g Two signaling protocols
  - H.323
  - SIP [Session Initiation Protocol]

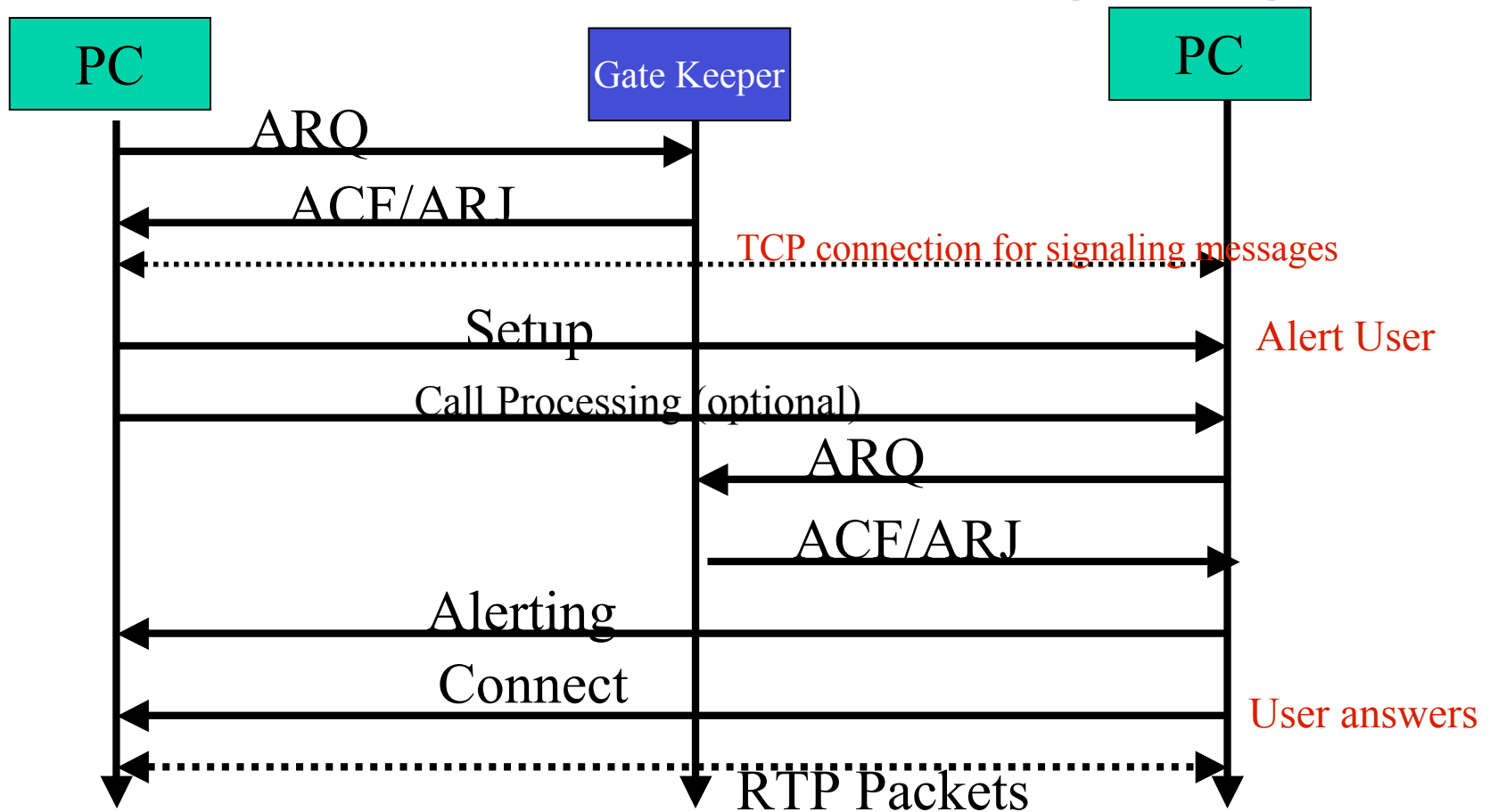
## Signaling [H.323]

- g End point: Entity making/receiving calls
- g Gate Keeper: Performs Address Translation, call control, etc. (one per LAN)
- g End point (in our case PC) registers with GateKeeper

## Registration



## Call Signaling



## DTMF: Dual Tone Multi-Frequency

- g OK with G.711 (coder with no assumption about signal being sound)
- g Other codecs: Cannot transmit DTMF's (Here accuracy is more important than timing)
- g Special messages carry IVR (interactive Voice Response) responses
- g Special RTP logical channel to carry DTMF

## SIP (Session Initiation Protocol)

- g Signaling protocol to
  - Initiate
  - Manage and
  - Terminate voice/video sessions on packet networks
- g  $\geq 1$  participants, unicast/multicast
- g Text coded and can be extended

## SIP entities

- g User agent – Initiates/terminates sessions [client initiates SIP request, server serves SIP request]
- g Proxy Server – Both as client and server, interprets requests, can serve locally, translate/rewrite and forward
- g Redirect Server – Accept SIP request, map to  $\geq 0$  new addresses and return to client
- g Registrar – [like HLR] accepts REGISTER request, update database

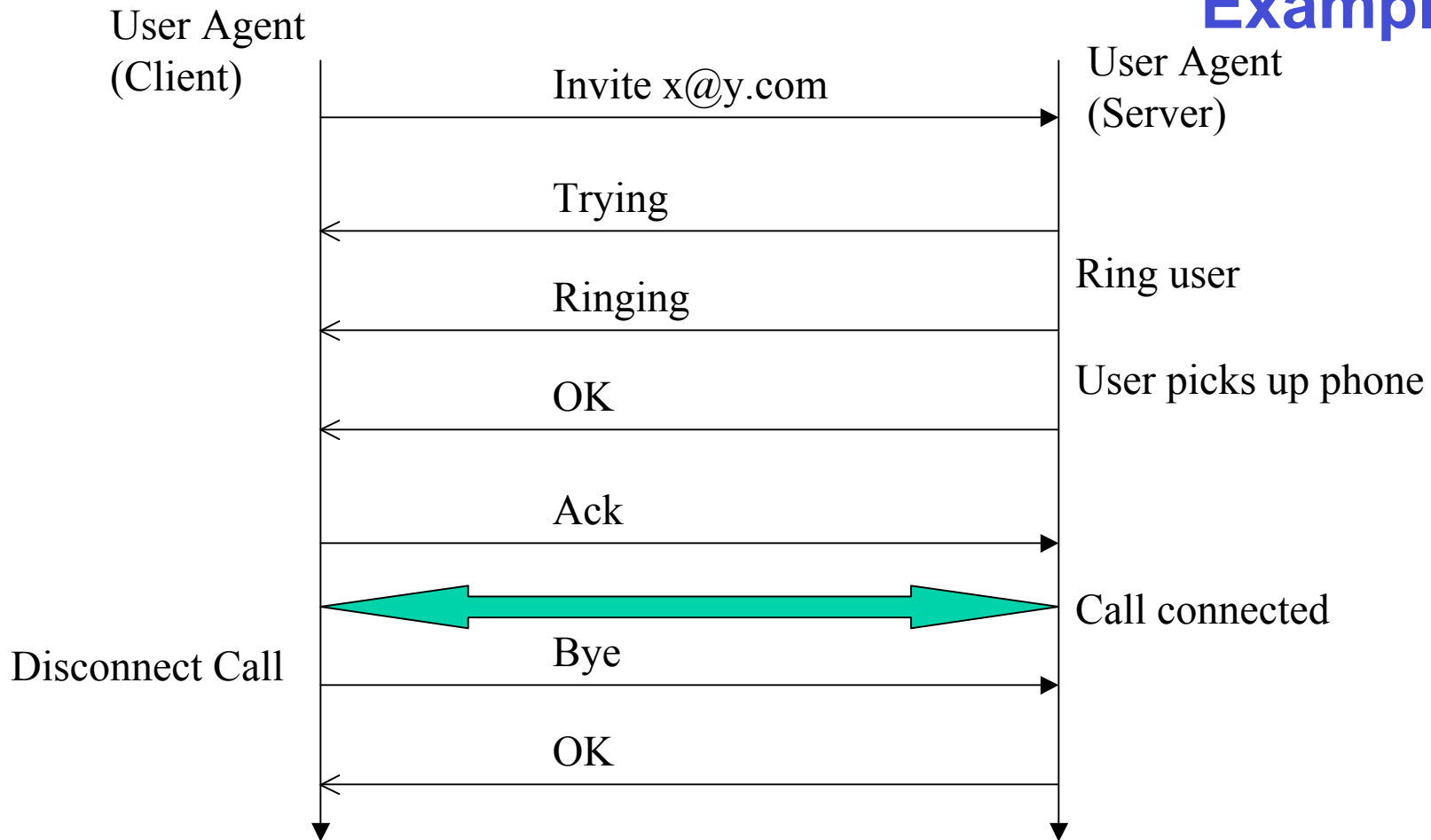
## SIP Messages

- g Request and response
- g SIP messages are sent over UDP/TCP
- g SIP is used with SDP (Session Description Protocol); SDP describes capabilities of the sender [vocoder used, etc.]

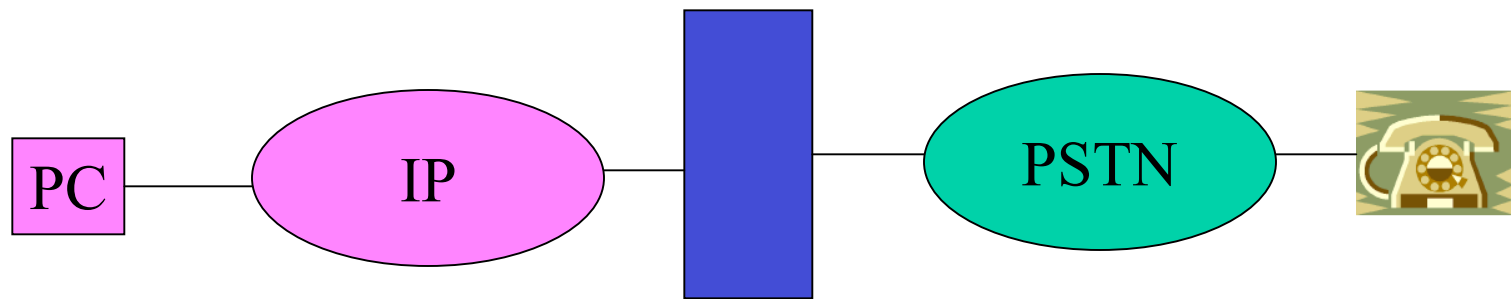
## SIP Messages

- g INVITE – initiate session; Session description included in message body
- g ACK – confirm session establishment
- g BYE – terminate connection
- g CANCEL – cancel pending INVITE
- g OPTIONS – capability inquiry
- g REGISTER – bind permanent address to current location

## Examples

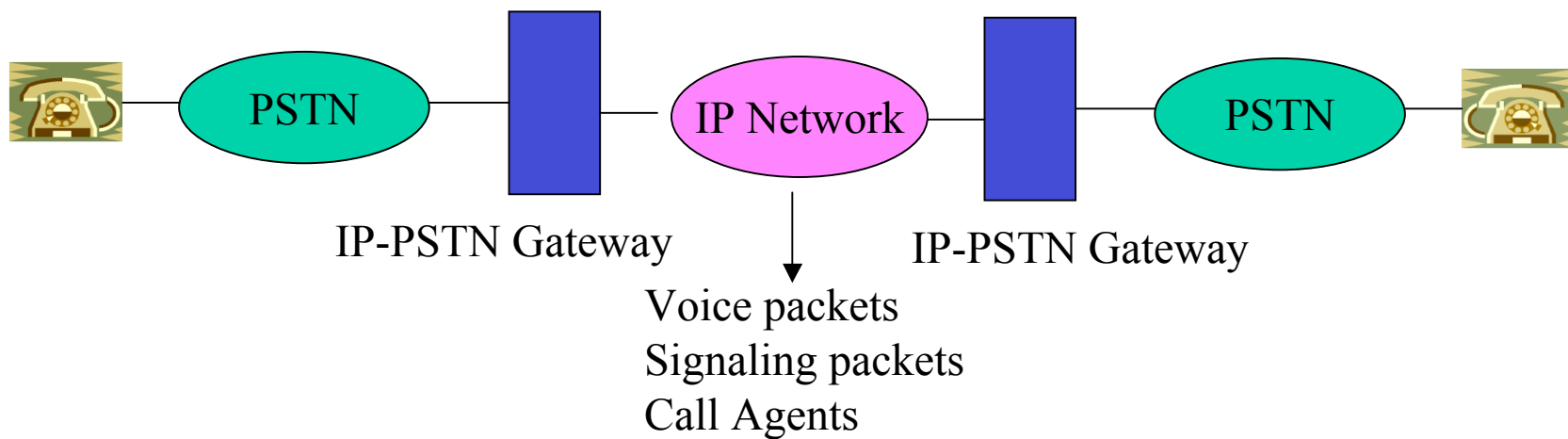


PC <-> PSTN?

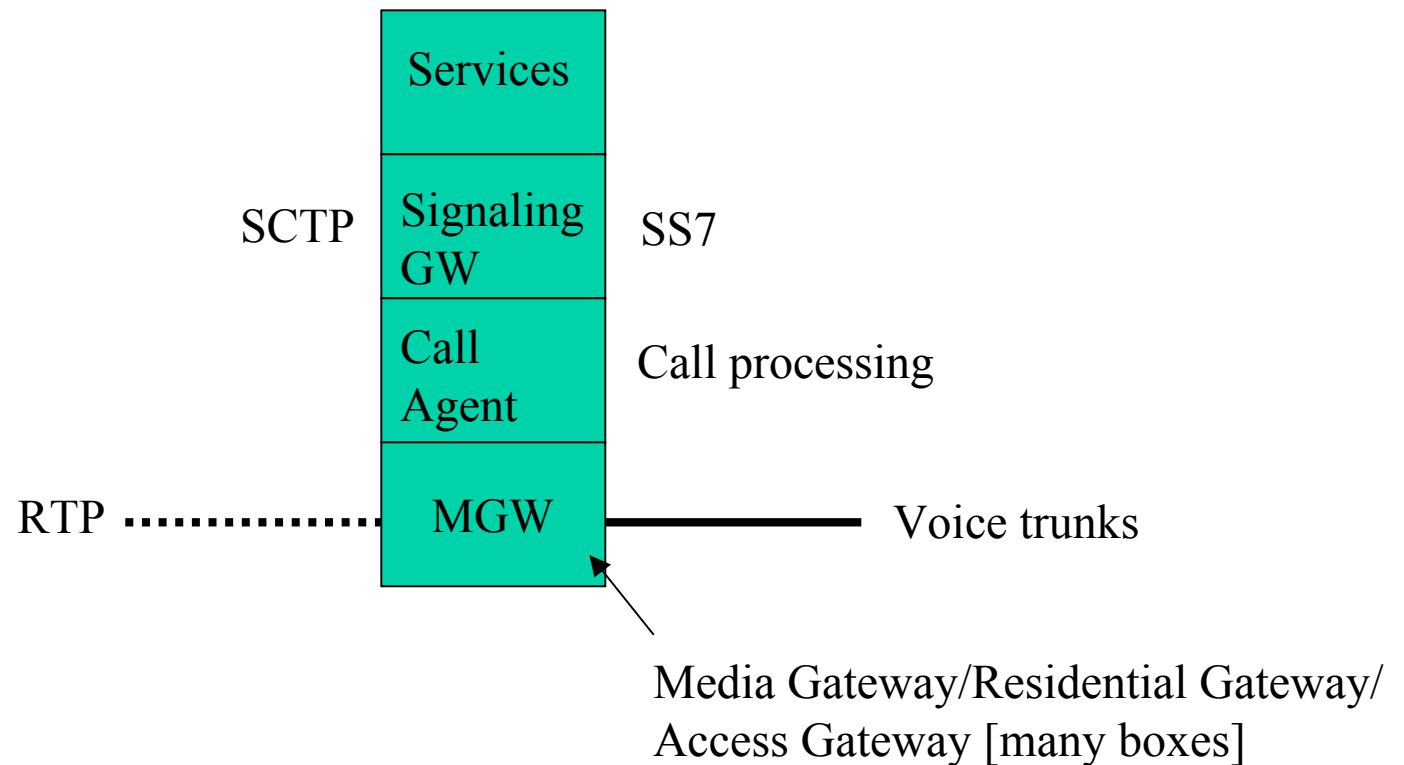


IP-PSTN Gateway (Voice GW and Signaling GW)

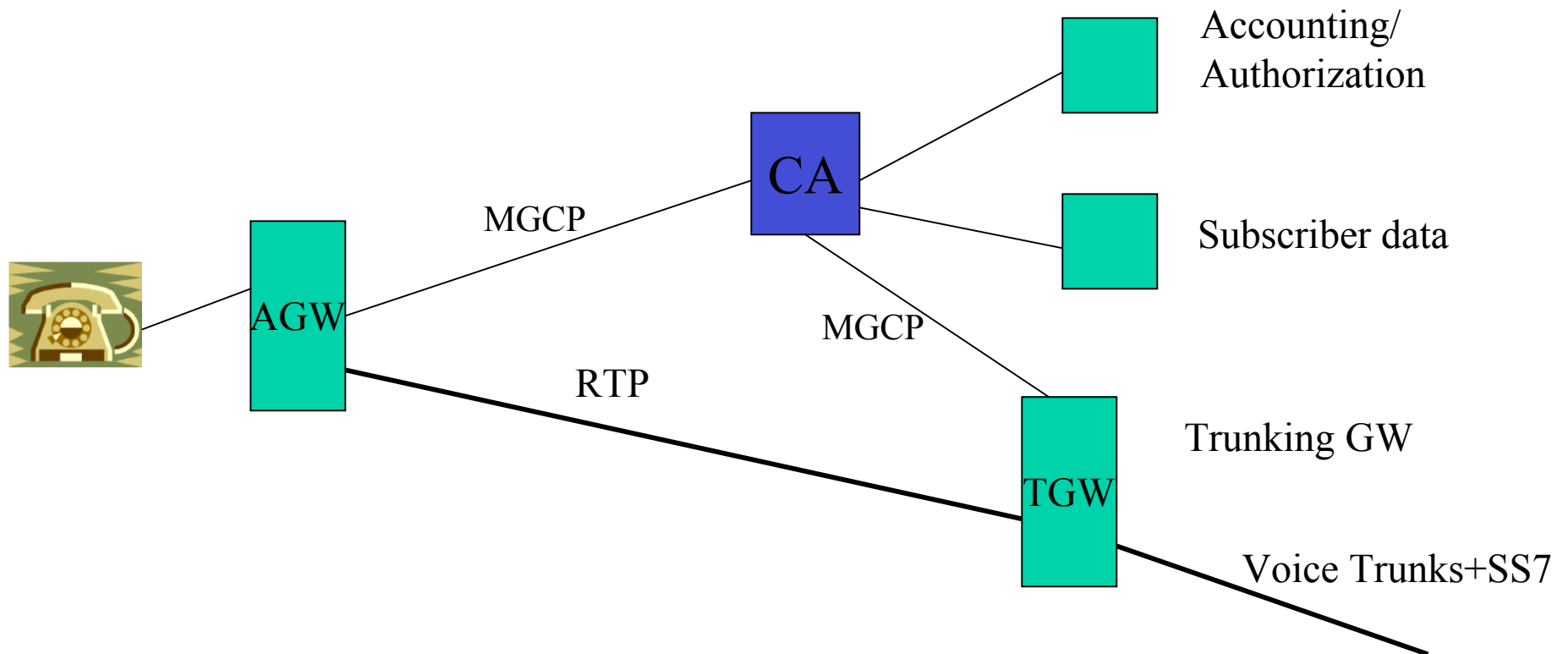
## PSTN <-> PSTN over IP?



## IP-PSTN Gateway



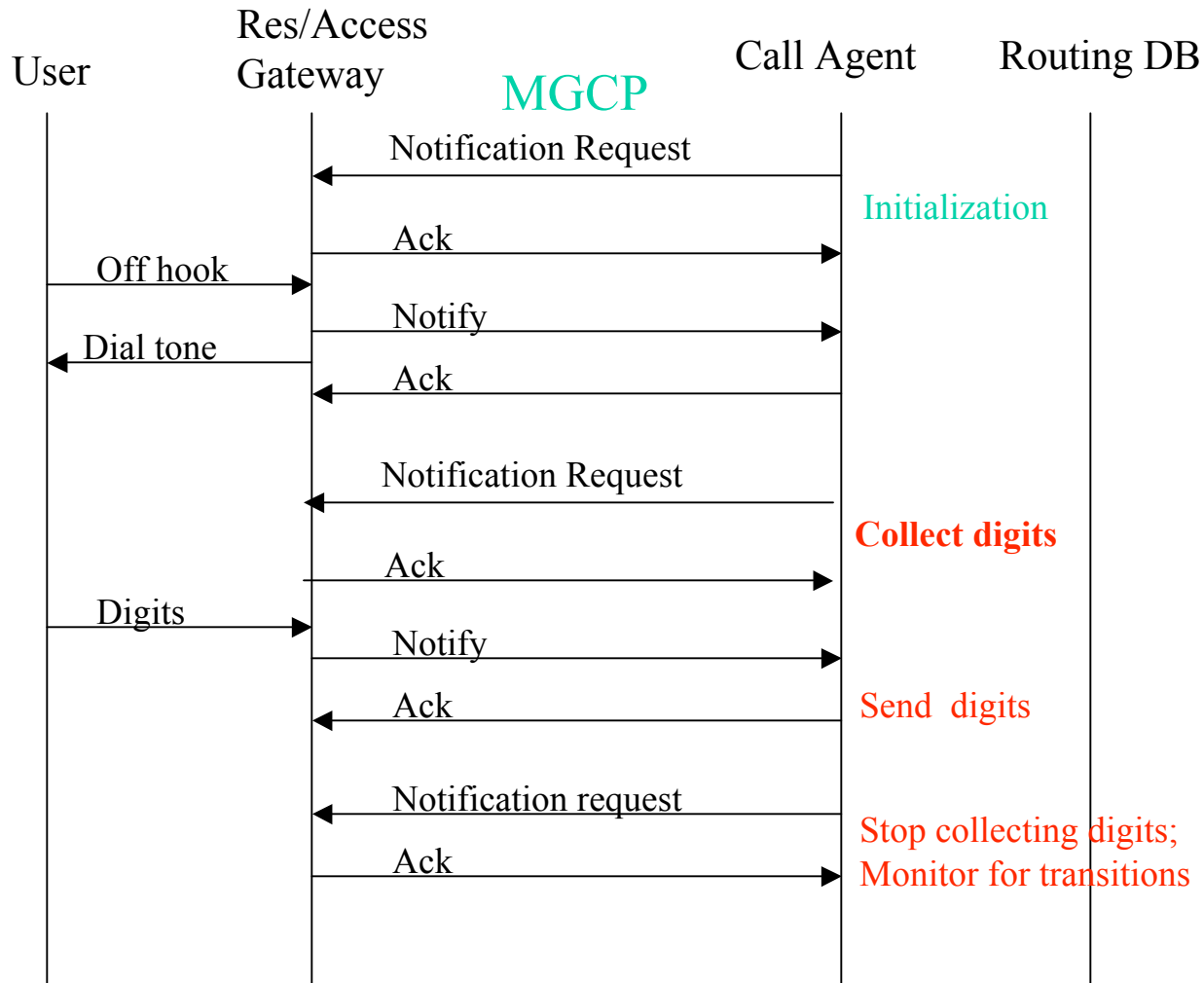
## Big Picture

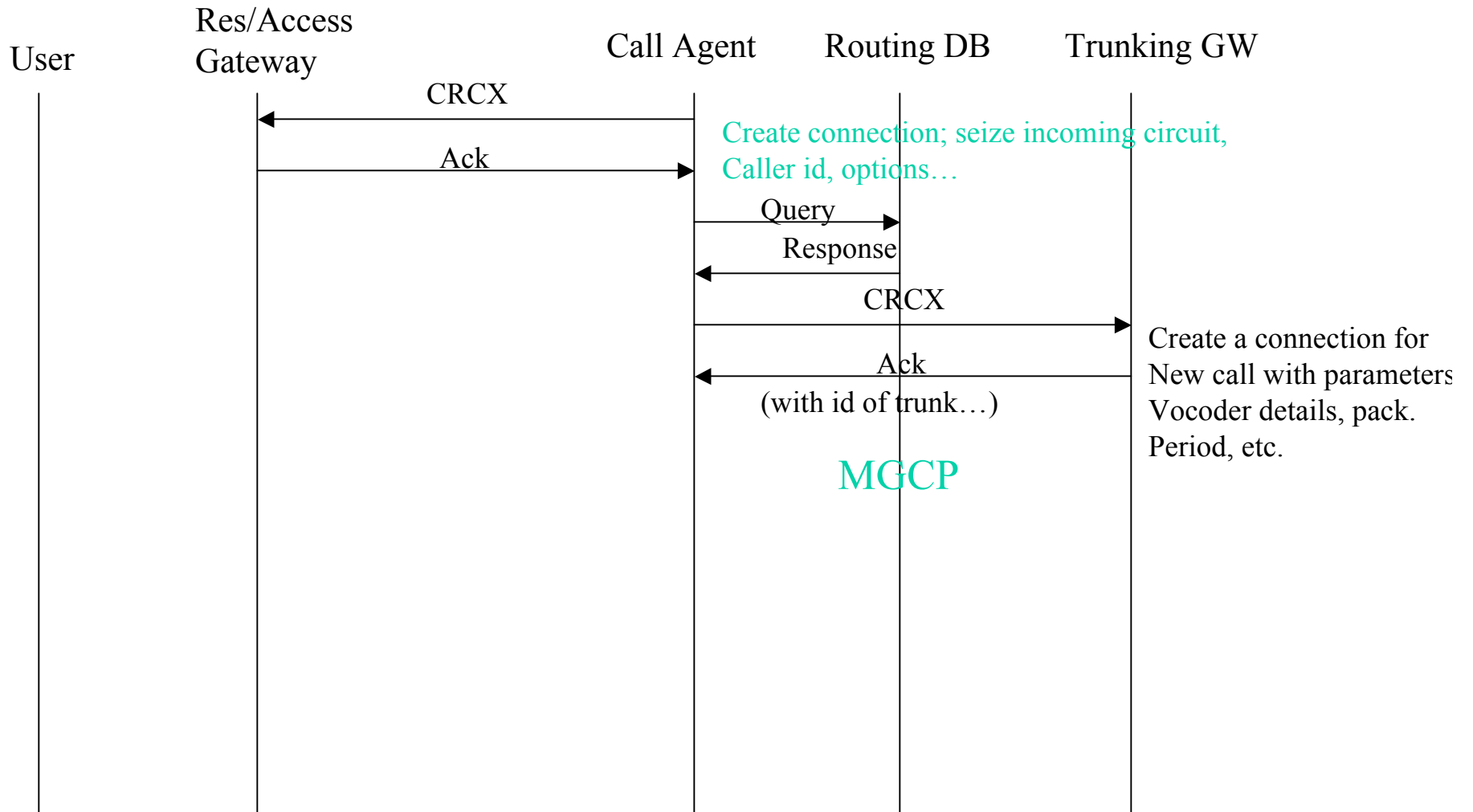


## MGCP [Media Gateway Control Protocol]

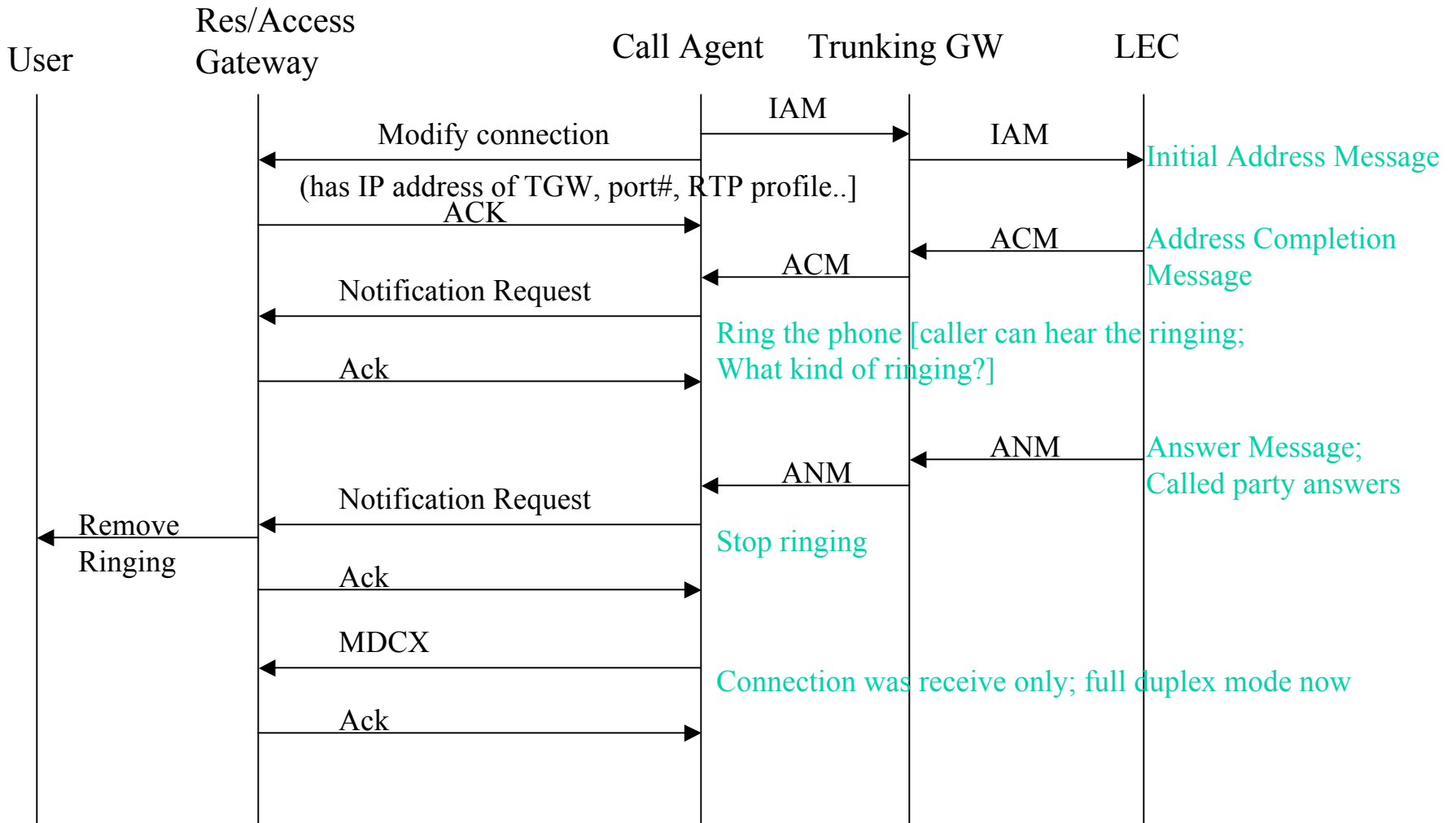
- g 8 MGCP commands:
  - Notification request [CA->GW]
  - Notify [tell CA when an event has occurred]
  - CRCX [create connection]
  - Modify Connection [change parameters of a previously created connection]
  - DLCX [Delete existing connection]
  - Audit endpoint/connection
  - Restart in progress [GW->CA; GW is going down or in process of coming up]

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