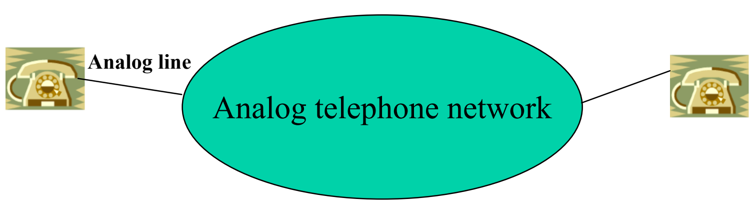
Introduction to VoIP

RFCs (RTP, SIP, H.323)

Various books on VoIP

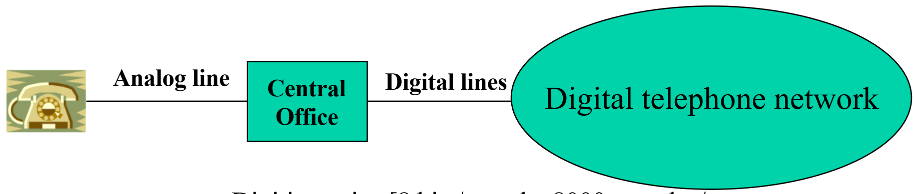
Telephony



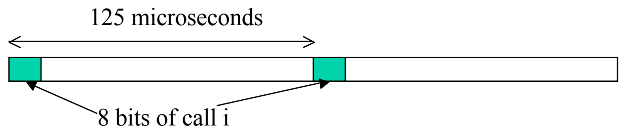
Manual operators on switchboard or
Electromechanical gear

Problems: Maintenance – nightmare
Noise – multiplex, diff voice circuits on diff freq

Digital Telephony



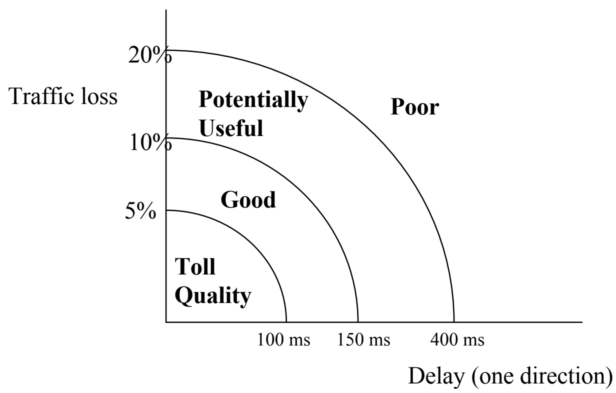
Digitize voice [8 bits/sample, 8000 samples/s
Multiplex voice samples



Statistical MUX/Asynchronous Transmission

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

Call Quality



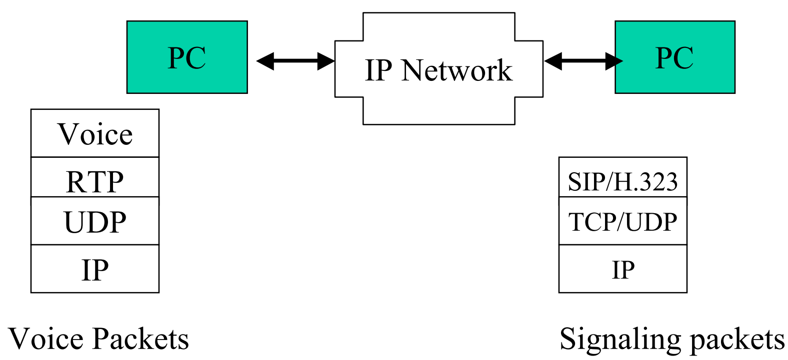
Need for VoIP

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

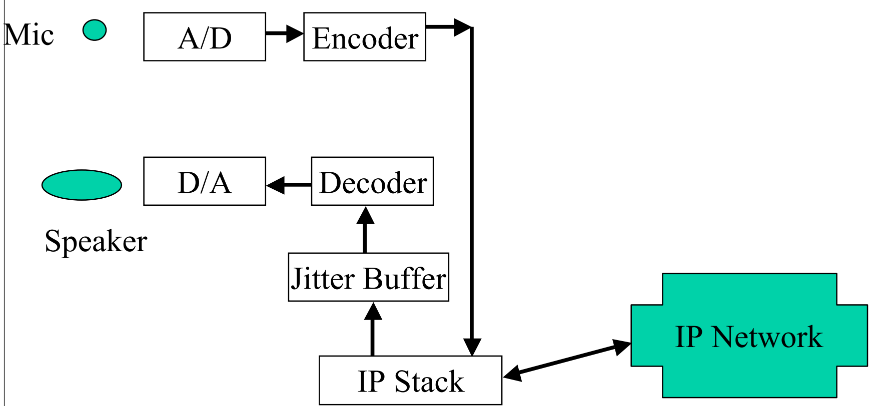
Main issues

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

Protocols Used (PC-PC call)



Architecture



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RTP (Real-Time Transport Protocol)

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

RTP

- Defines a way to format IP packets carrying time-sensitive data
- 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)

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Payload Type

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

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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

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

RTP Header Cont'd

- Seq #: start at random number; increment by 1 for each packet
- time stamp[32 bits]: time value when first sample in payload was sampled;
  - initial value chosen randomly
- SSRC: ID of source of RTP stream
- 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:

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

RTCP Header

V (2)	P(1)	count (5)	Type (8)	Length (16)
Data				

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

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RTCP Header

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

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RTCP Header Cont'd

- 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

- Si = Time Stamp of sender of packet i
- Sj = Time Stamp of sender of packet j
- Ri = Time Stamp of receiver of packet i
- Rj = Time Stamp of receiver of packet j
- D[i,j]= (Rj-Ri)-(Sj-Si): Interarrival jitter
- Cumulative Jitter J = J+(|D[i-1,j])|-J)/16

Echo Handling

- Traditionally, 4 wire to 2 wire transformation adds echo

- Two types of echo:
  - Electric echo
  - Acoustic echo

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Acoustic Echo

- Time between speech and the same sound getting into the mic:
  - <=20 milliseconds? No noticeable echo
  - >=40 ms? Problem
  - Speaker phone is also a source of echo

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Echo Suppressers:

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

Echo Cancellers

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

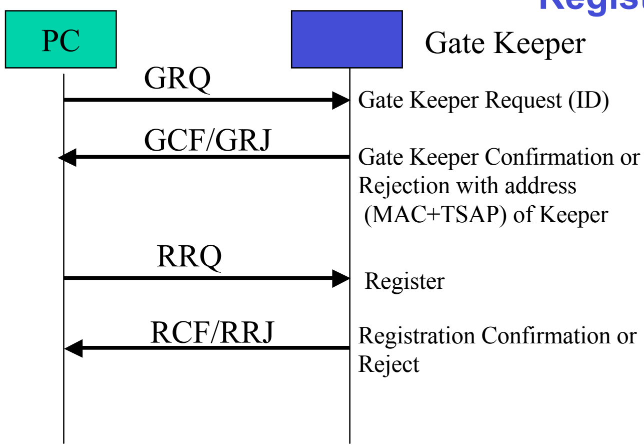
Signaling

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

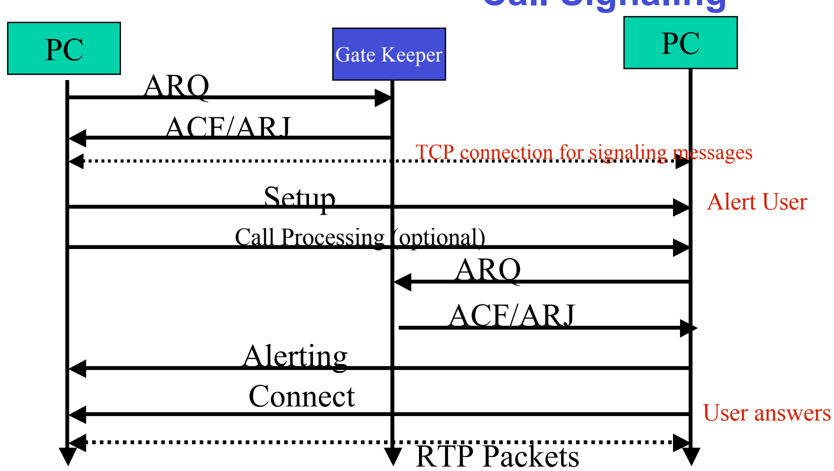
Signaling [H.323]

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

Registration



Call Signaling



DTMF: Dual Tone Multi-Frequency

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

SIP (Session Initiation Protocol)

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

SIP entities

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

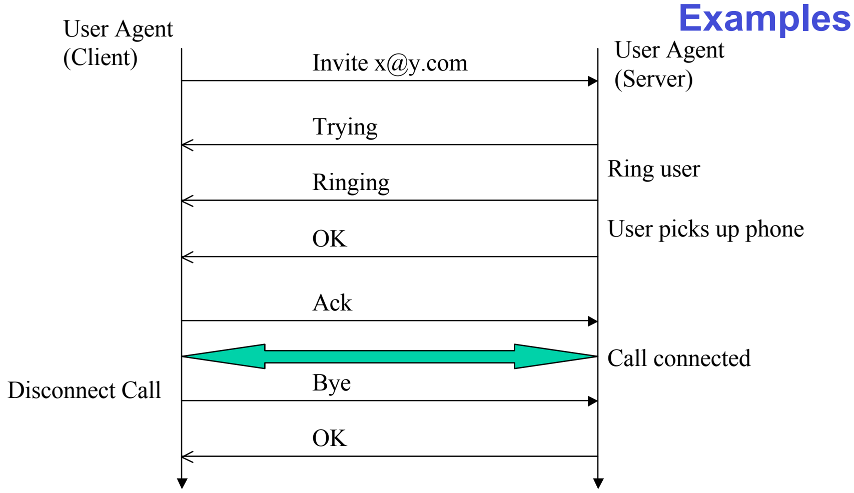
SIP Messages

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

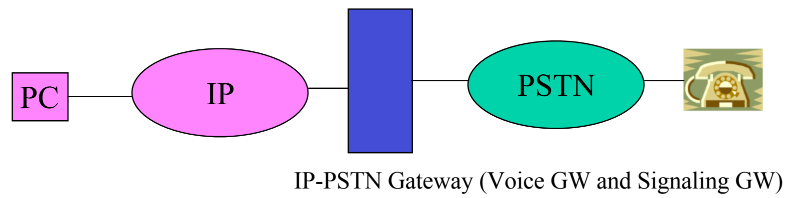
SIP Messages

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

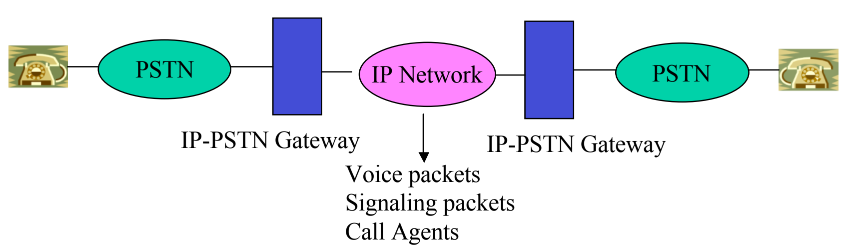
Examples



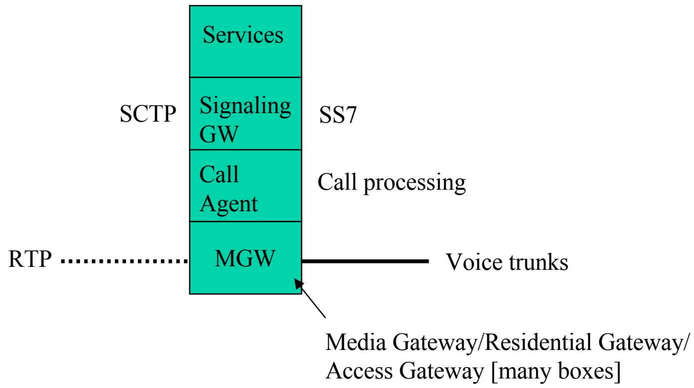
PC <-> PSTN?



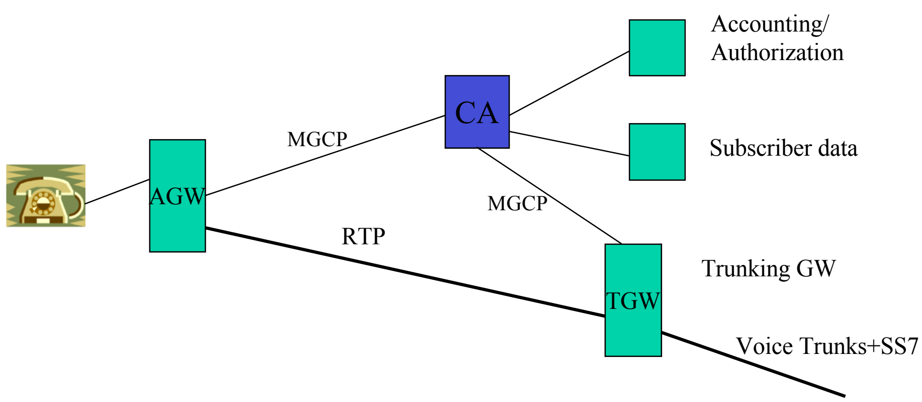
PSTN <-> PSTN over IP?



IP-PSTN Gateway

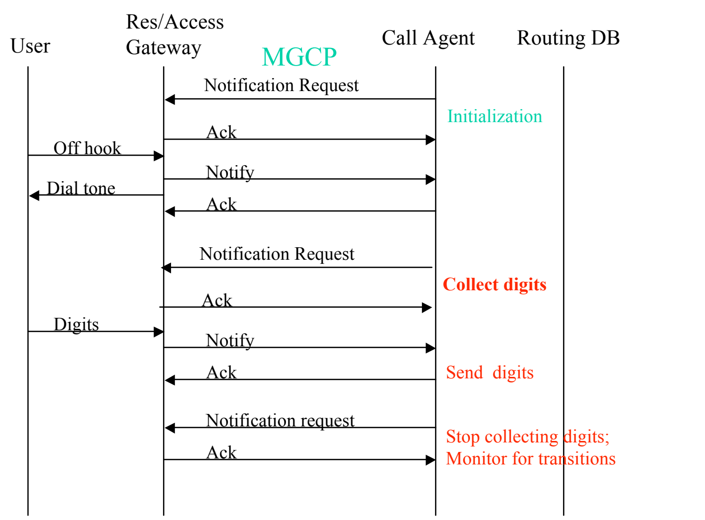


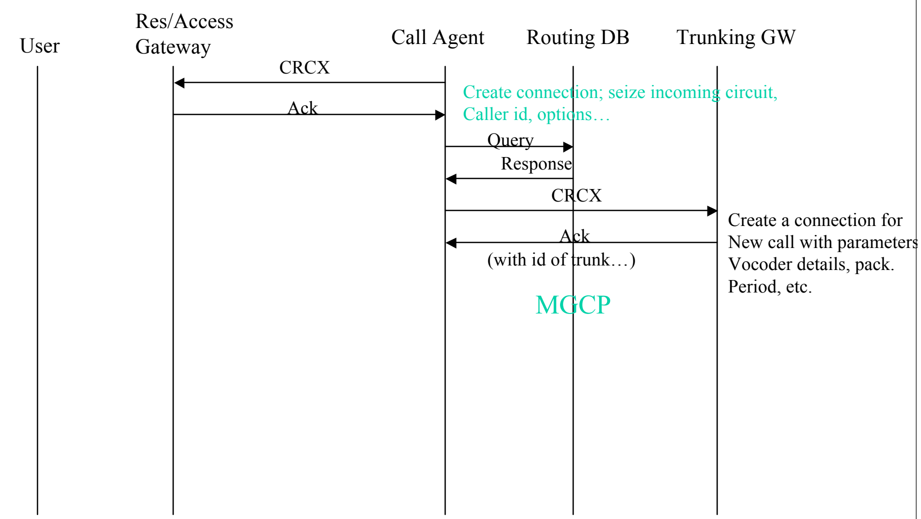
Big Picture

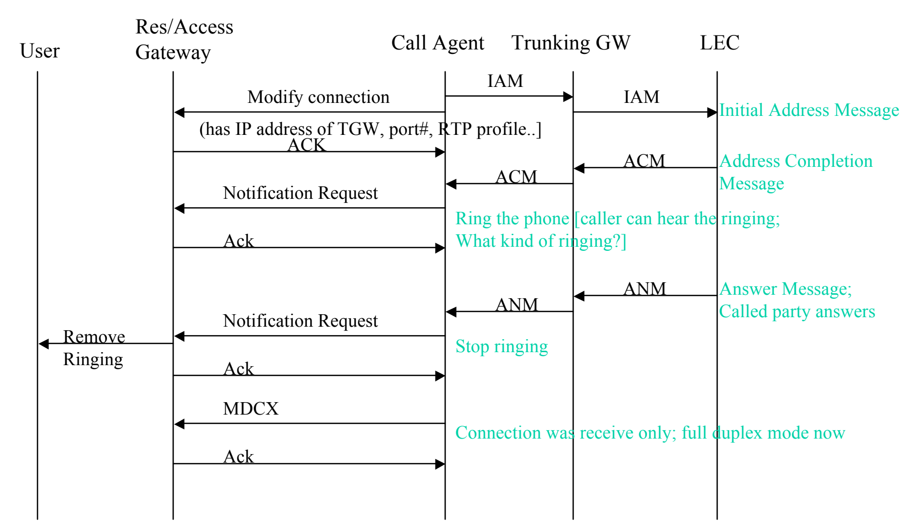


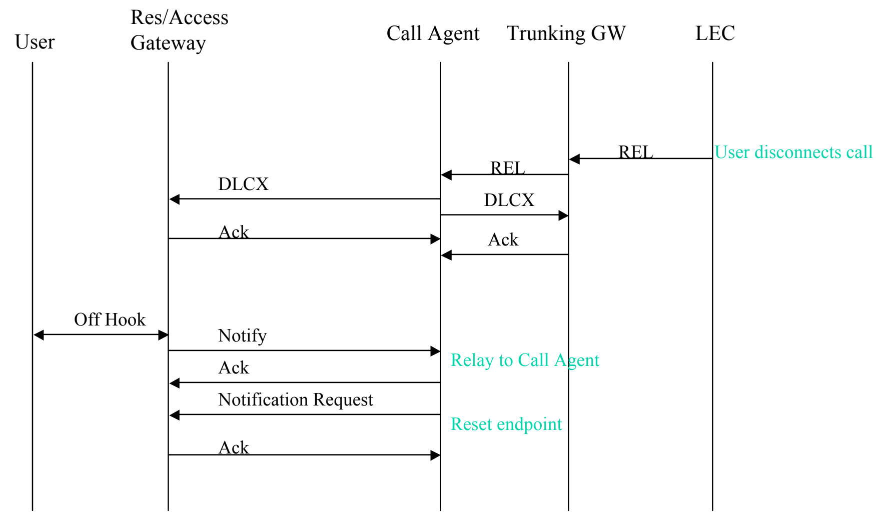
MGCP [Media Gateway Control Protocol]

- 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]




User
Res/Access Gateway
Call Agent
Routing DB
Trunking GW
CRCX
Ack
Create connection; seize incoming circuit, Caller id, options…
Query
Response
CRCX
Ack
(with id of trunk…)
Create a connection for New call with parameters Vocoder details, pack. Period, etc.
MGCP


User
Res/Access Gateway
Call Agent
Trunking GW
LEC
IAM
IAM
Initial Address Message
Modify connection
(has IP address of TGW, port#, RTP profile..]
ACK
ACM
ACM
Address Completion Message
Notification Request
Ring the phone [caller can hear the ringing; What kind of ringing?]
Ack
ANM
ANM
Answer Message; Called party answers
Notification Request
Remove Ringing
Stop ringing
Ack
MDCX
Connection was receive only; full duplex mode now
Ack


User
Res/Access Gateway
Call Agent
Trunking GW
LEC
REL
User disconnects call
REL
DLCX
DLCX
Ack
Ack
Off Hook
Notify
Relay to Call Agent
Ack
Notification Request
Reset endpoint
Ack