VOICE OVER CIRCUIT SWITCHING NETWORKS

PHASE TWO - ANALOG TELEPHONE, DIGITAL EXCHANGE

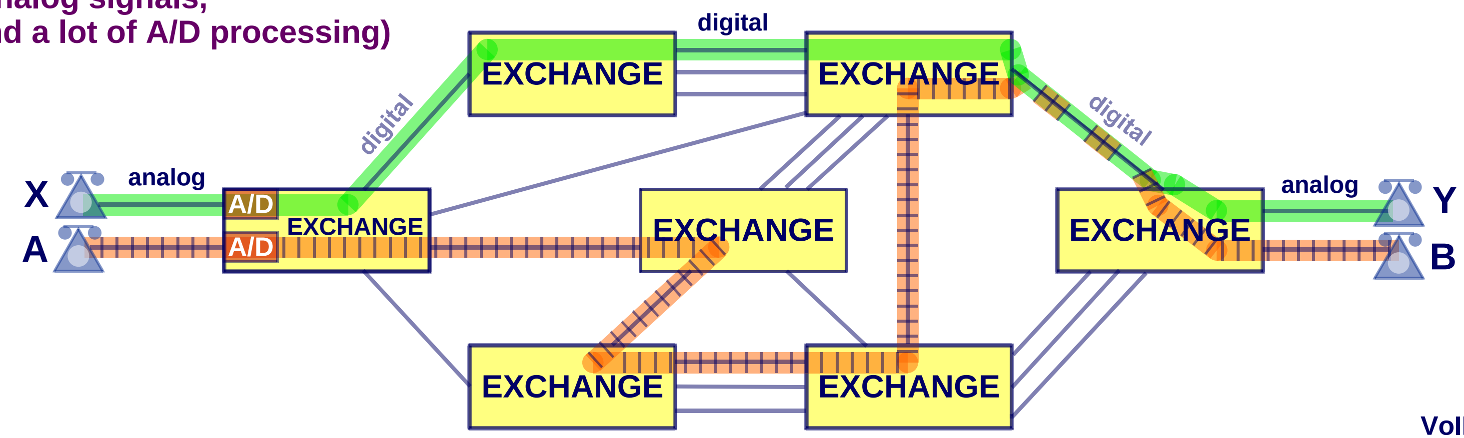
- Signals - User to exchange: analog voice
- Exchange to exchange: digitized voice
- Technology - Virtual circuit switching and multiplexing

<u>Advantages:</u>

- **Switching is faster (electronic)**
- **Circuits are "logical" not "physical". Multiple logical (virtual) circuits may be multiplexed over one single high speed trunk**
  24 x 64 kbps ⇒ 1.544 Mbps (T1)
  32 x 64 kbps ⇒ 2.048 Mbps (E1)
- **As most of the circuit carries digital signals, noises have less influence ⇒ better voice quality**

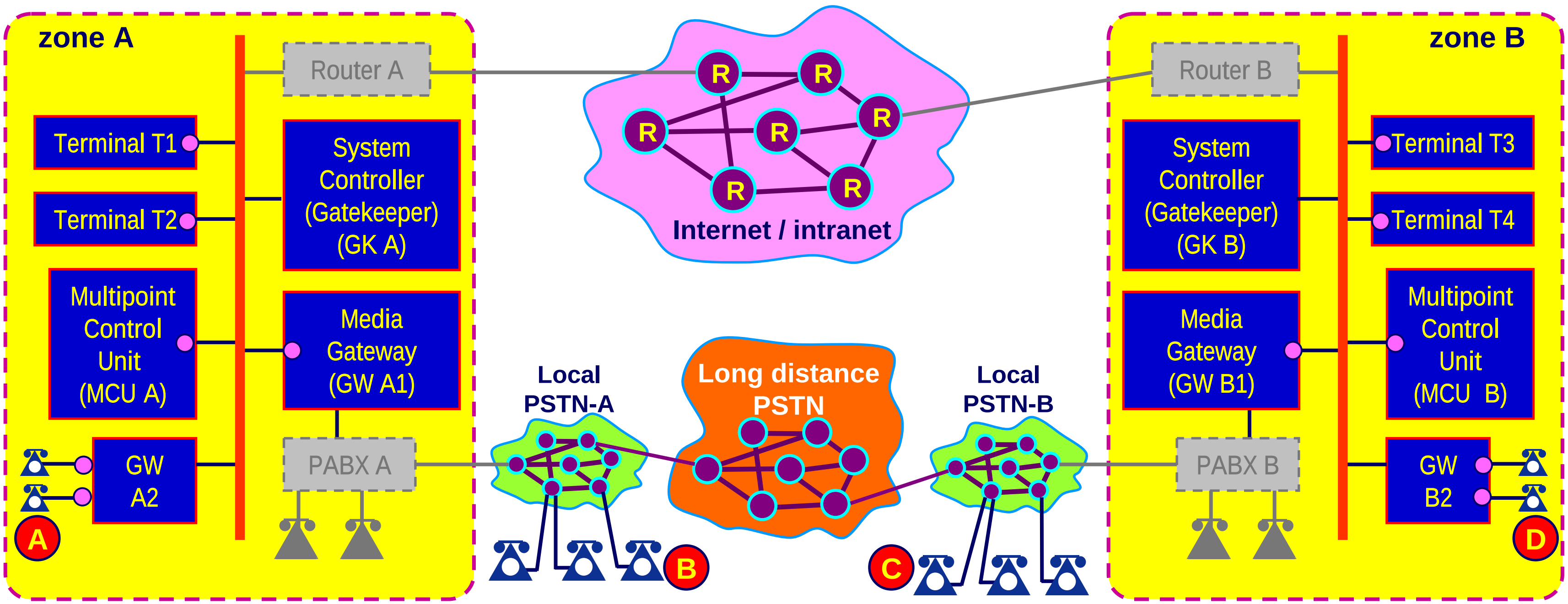
<u>But:</u>

- **Two phones still require two separate physical lines! - No multiplexing in the local loop.**
- **Digital signals (generated by PCs, smoke detectors, TV cameras, etc.) have to be <u>downgraded</u> to analog signals just to be digitized again in the exchange, to be converted to analog at the remote site, to be converted back in digital format by the final device!**
- **Lines are noisy (analog signals, and a lot of A/D processing)**



VoIP CALL FLOW

VoIP DEVICE to VoIP DEVICE



In different zones

- *T1 to T3*
- GK B finds “tel-(B)T3” in its table and sends to GK A, “IP-T3”
- GK A sends to T1, “IP-T3”
- T1 generates VoIP packets with IP DA = IP-T3

GK B Table

Phone number	IP address
tel-(B)T3	IP-T3
tel-(B)T4	IP-T4
tel-(972)…(e.g.B)	Zone A; IP-GK A
tel-(B)…(e.g.D)	IP-GW B2
tel-(86)…(e.g.C)	IP-GW B1
tel-(A)…(e.g.A,T1,2)	Zone A; IP-GK A

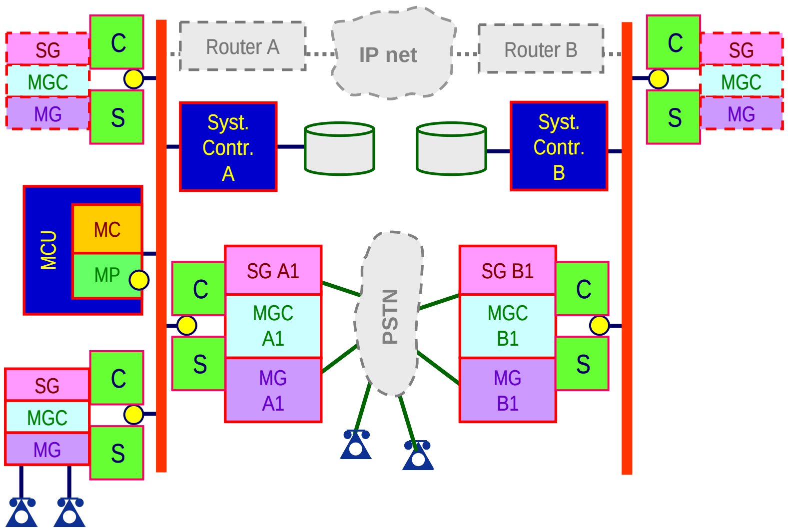
GK A Table

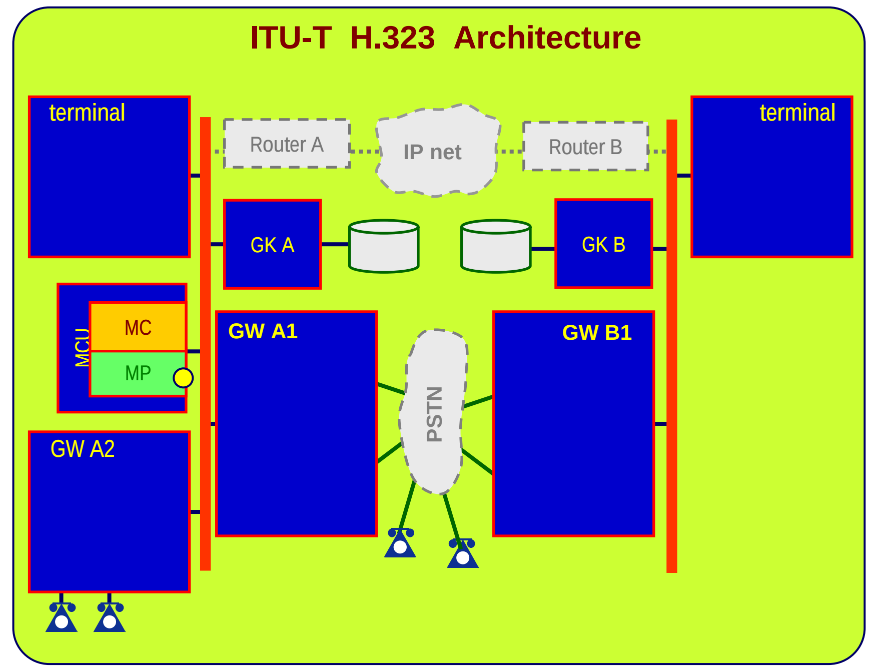
Phone number	IP address
tel-(A)T1	IP-T1
tel-(A)T2	IP-T2
tel-(972)…(e.g.B)	IP-GW A1
tel-(A)…(e.g.A)	IP-GW A2
tel-(86)…(e.g.C)	Zone B; IP-GK B
tel-(B)…(e.g.D,T3,4)	Zone B; IP-GK B

VoIP SYSTEM ARCHITECTURE

ITU-T H.323

Summary

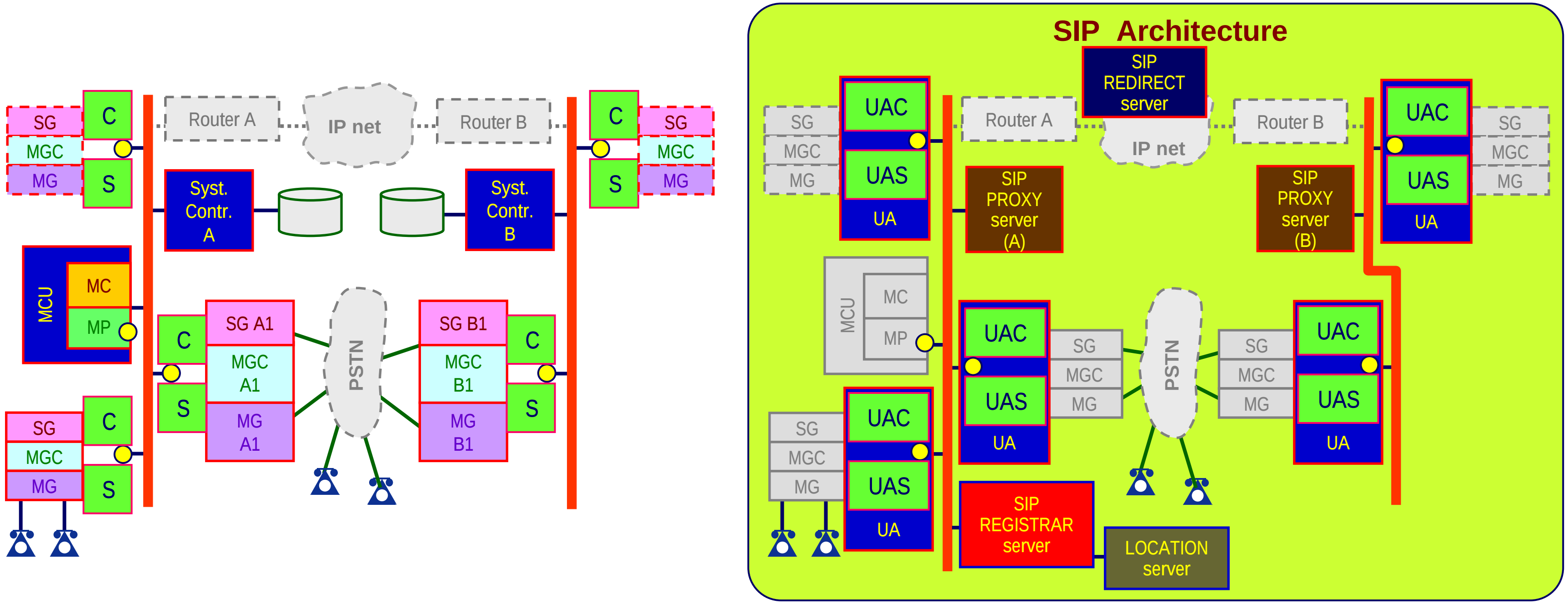




H.323 basic call flow

1.- End points register with the zone GK
2.- Terminal / GW query GK for IP address to be used for controls
3.- GK may indicate as the entity to send controls to:
- itself (GK routed call), or
- the remote end (direct routed call)
- If GK has no relevant information, it may contact other servers (local or in the Internet)
4.- Terminal / GW query the entity indicated by GK in step 3, for the IP address to be used to send media

VoIP SYSTEM ARCHITECTURE
SIP (Session Initiation Protocol)

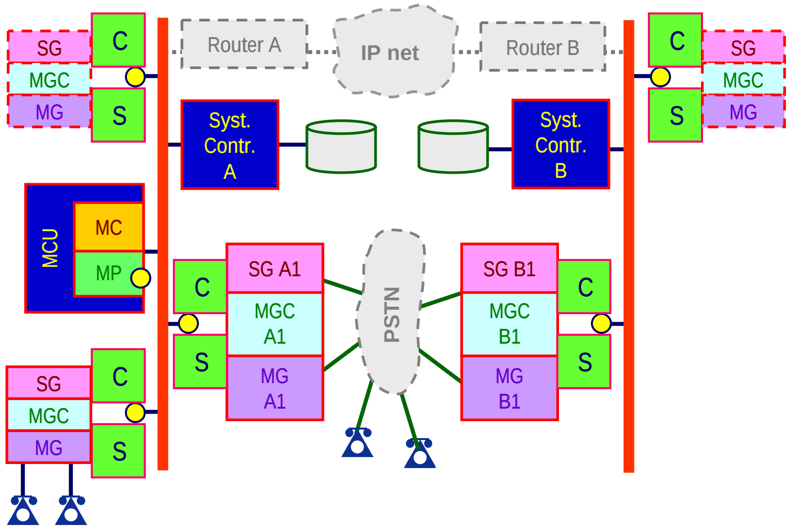


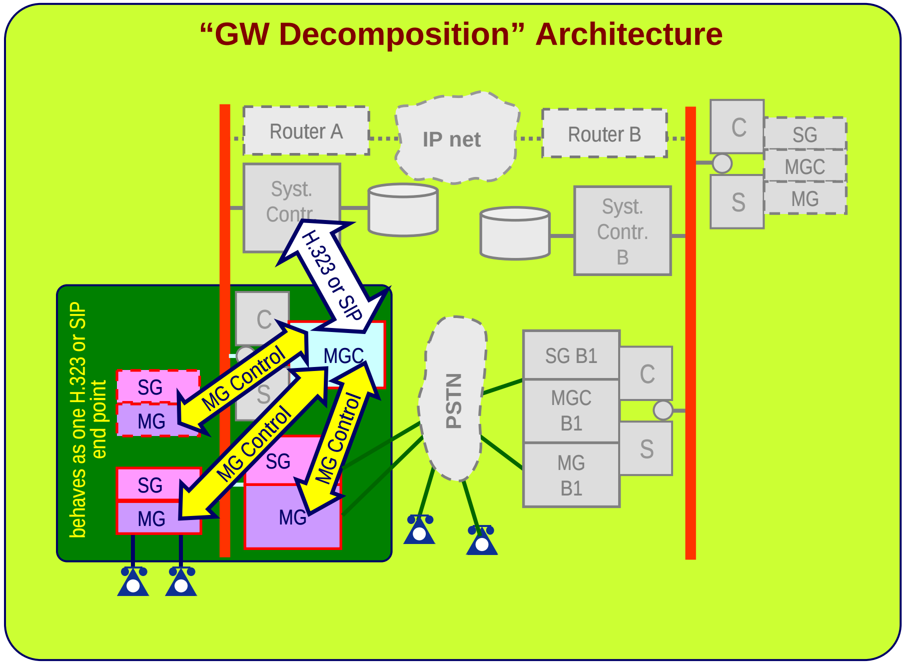
SIP basic call flow

1.- User Agents register with the Registrar server (Registrar server maintains user's whereabouts in a Location Server
2.- User Agent Client query SIP server for IP address to be used for controls
3.a.- Proxy server indicates itself as entity to send controls to.
It forwards client's requests to the called party or to a better informed server (it acts on client's behalf).
It executes call signaling on behalf of the party it serves (retaining billing information).
3.b.- Redirect server responds to client's request by providing to it the coordinates of the called party or those of another server, better informed about called party location. Client has to contact directly the new server.
3.c.- Last server has to indicate the coordinates of the called party (for media)

VoIP SYSTEM ARCHITECTURE

Gateway Decomposition

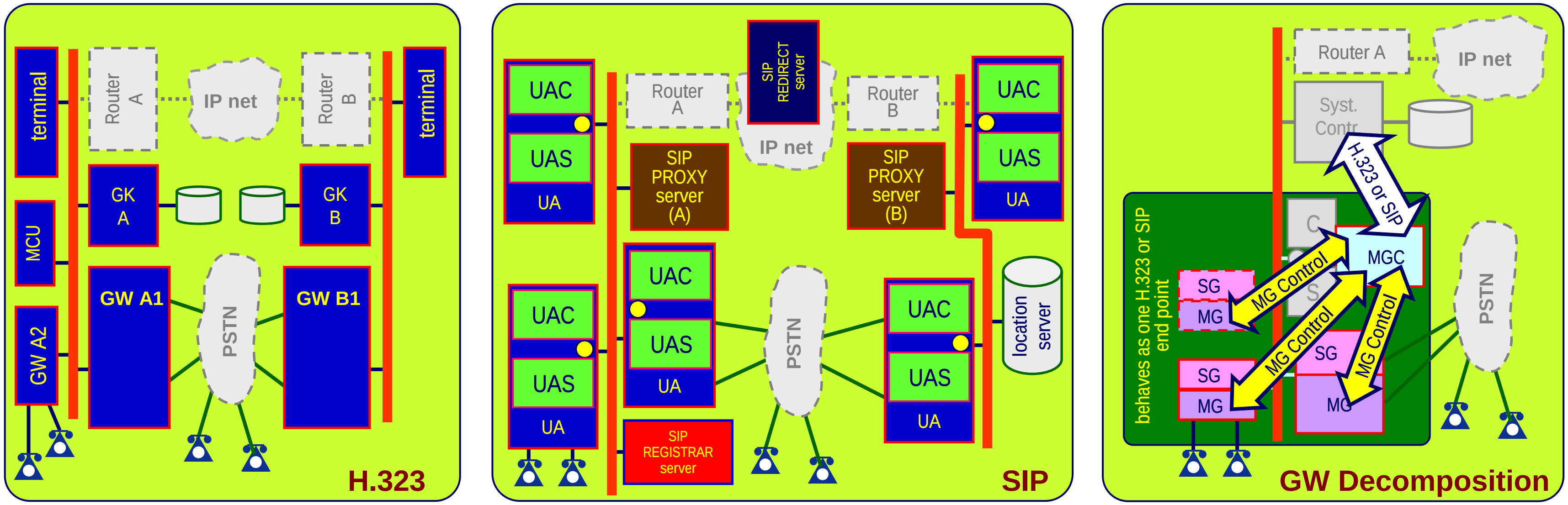




- The GW is decomposed into its basic elements:
    - one Media Gateway Controller (MGC)(a.k.a Agent, Call Agent)
    - multiple Media Gateways (MG)
- MG reports to MGC every event (incoming call, off hook, hang up, etc.).
- In response, MGC instructs MG what has to be done (ring the phone, bring dialed numbers, etc.).
- MG is simple (cheap)
- Future services are transparent to MG. Changes affect only MGC, which will generate new sets of commands to MG, in order to provide the new services
- For the external world, MGC together with all its MGs looks like a node with multiple connections
- MGs are unaware that the call is established by MGC using H.323 or SIP.
- Only MGC understands both MGCP and SIP or H.323

VoIP STANDARDS

MAIN STANDARDS



	Field	ITU-T SG16 (H.323 suite)	IETF (SIP suite)	Telcordia (Bellcore)	ETSI (TIPHON)
Hand shaking processes	Registration	RAS	SIP	(-)	Restrictions on H.323 usage: DTS 2001 - PC to phone DTS 2002 - phone to PC DTS 2003 - phone to phone DTS 2004 - PC to PC
	Call signaling	H.225.0 (Q 931)	SIP	(-)	
	Capabilities exchange	H.245	SIP (SDP)	(-)	
	Media GW Controller (MGC) to Media GW (MG)	- Internal to GW - H.248 (defined with IETF MEGACO) - MGCP	- MEGACO (defined with ITU-T SG16 H.248) - MGCP	MGCP	
	System controller to servers	not defined	not defined	(-)	