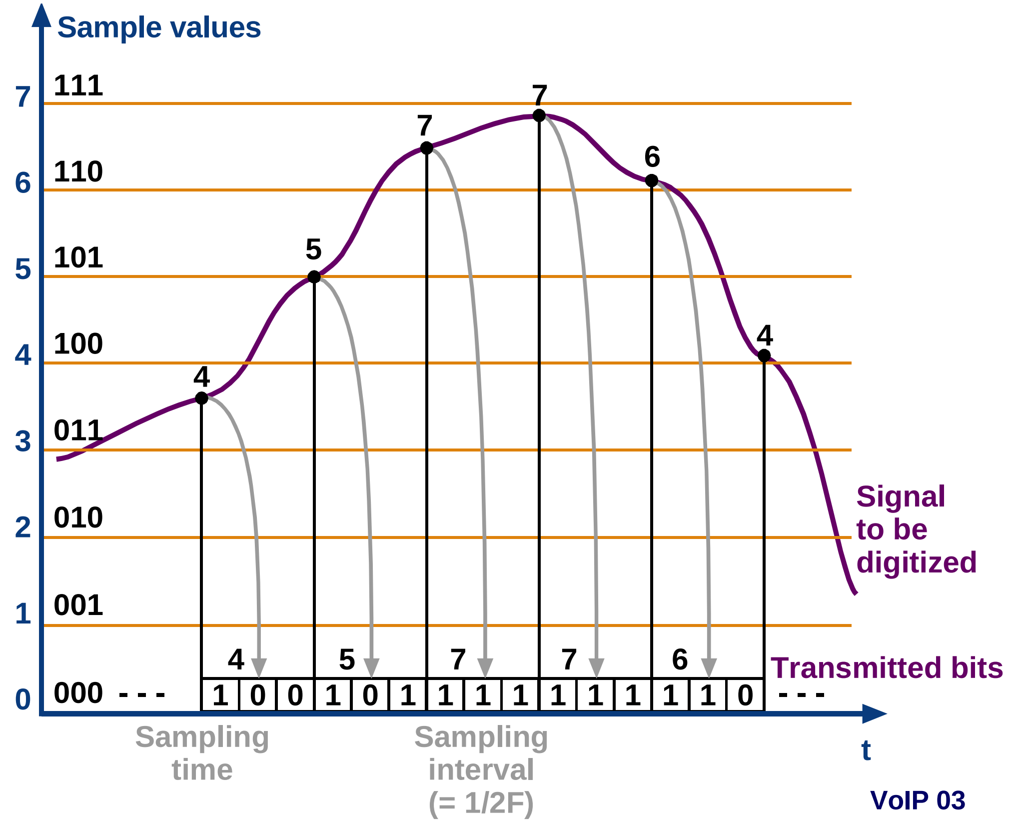
VOICE OVER CIRCUIT SWITCHING NETWORKS

DIGITIZING ANALOG SIGNALS

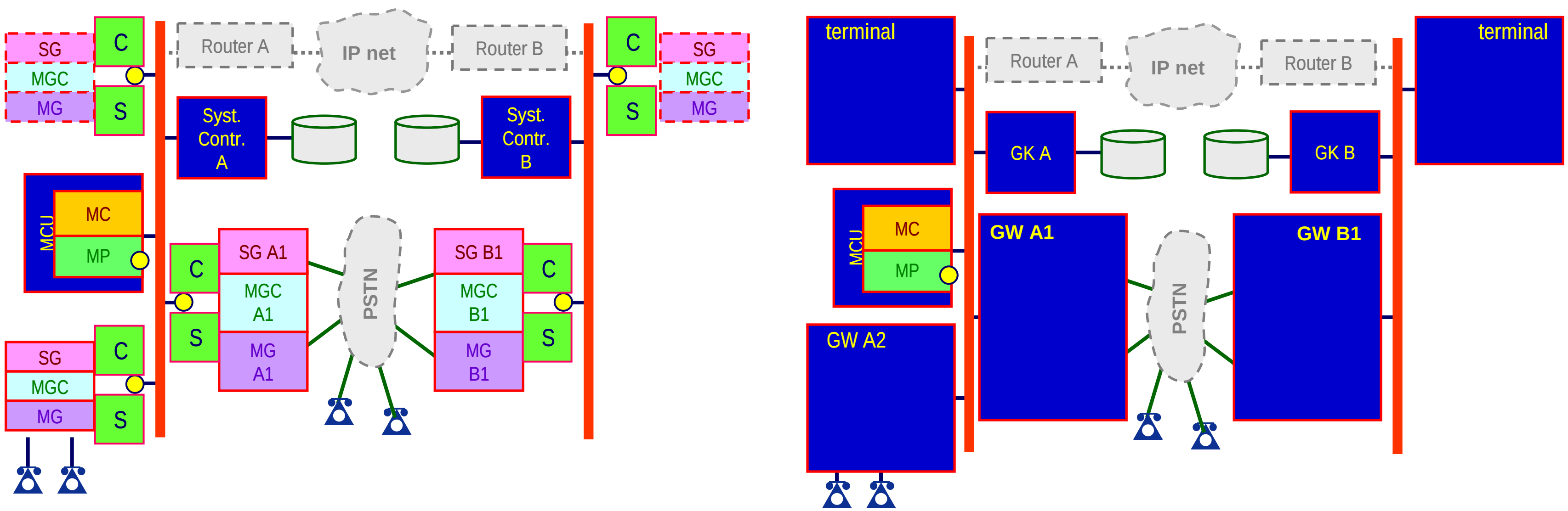
- H. Nyquist sampling theorem (1928)
  - A signal may be represented without loss of information, based only on samples taken from the signal at regular time intervals (= sampling rate)
  - For a signal with highest frequency component FHz, the sampling rate (for reproduction without loss of information) should be 2F Hz (= Nyquist sampling rate) or higher

- Every sample is represented by a number indicating its size.
- The number associated with a specific sample is represented in digital format (= code word). (The greater the number of bits representing the sample size, the more accurate the system is).
- The bits representing the number that represents the sample size (=code word) are transmitted in the time elapsing from one sampling moment to the next sampling moment.



SYSTEM ARCHITECTURE

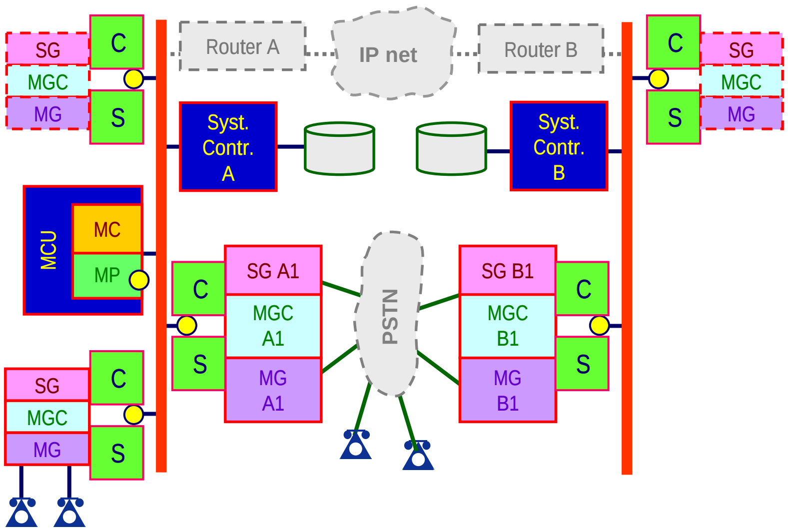
ITU-T H.323

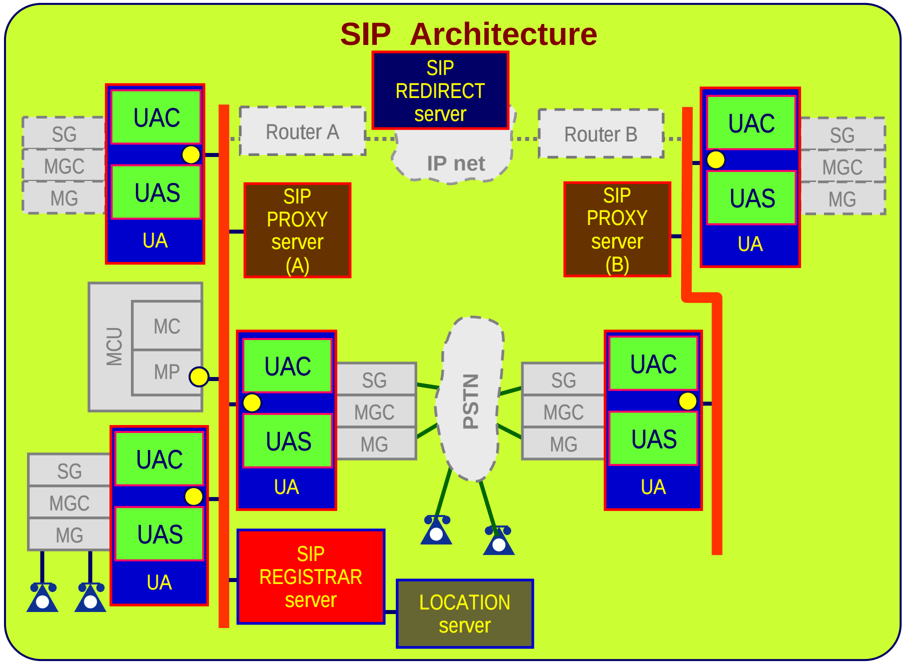


End Points	Terminal	T	T	Converts media into VoIP packets
End Points	Media Gateway	GW	MG	Converts media signals into VoIP packets
End Points	Signaling Gateway	GW	SG	Converts signaling into VoIP packets and commands to MG
End Points	Media Gateway Controller	GW	MGC	Controls MG based on signaling received by SG
End Points	Multipoint Processor	MCU	MP	Converts media types in multiparty sessions
Multipoint Controller		MCU	MC	Controls multiparty sessions
System Controller		GK	Syst Cont	Admission control, b/w management, IP-to-PSTN add conversion, etc.

VoIP SYSTEM ARCHITECTURE

SIP (Session Initiation Protocol)





Basic entities

- User Agent (UA)
    - User Agent Client (UAC) - initiates SIP requests
    - User Agent Server (UAS) - receives and responds to SIP requests on behalf of clients
- Servers (could be physically in same or different machines)
    - proxy
    - redirect
    - registrar
    - (location)

VoIP STANDARDS

MEDIA TRANSPORT STANDARDS

- ***RTP / RTCP (RFC 1889)***
  Real time Transport Protocol
  Real Time Control Protocol

- RTP

Field	Size	Comments
Ver.	2 bits	“VERSION”
P	1 bit	“PADDING” If P=1, payload is less than full packet (I.e. padding has been added); Last byte of padding indicates the amount of padding bytes that have been added / have to be ignored. Could be necessary in encryption algorithms using fix block size.
X	1 bit	“EXTENSION” If X=1, the packet includes exactly one RTP extension header (2 bytes)

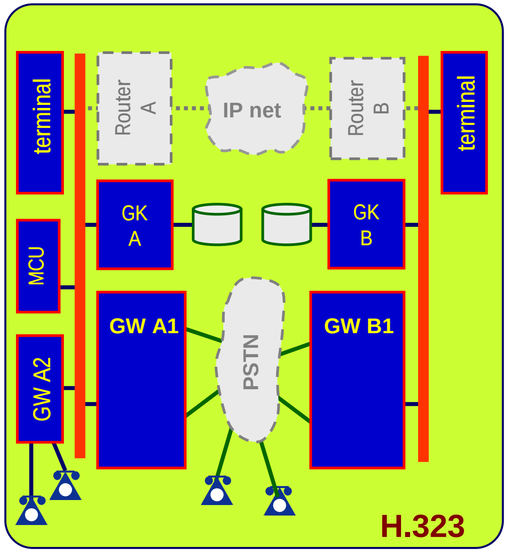
MAC header	(8)
IP header	20
UDP header	8
ver | P | X | CC	1
M | PT	1
Sequence Number	2
Time Stamp	4
SSRC	4
CSRC	(0-15)*4
Header Extension	(...)
Payload	(...)
padding	(...)
nr of padding bytes	1
MAC header	

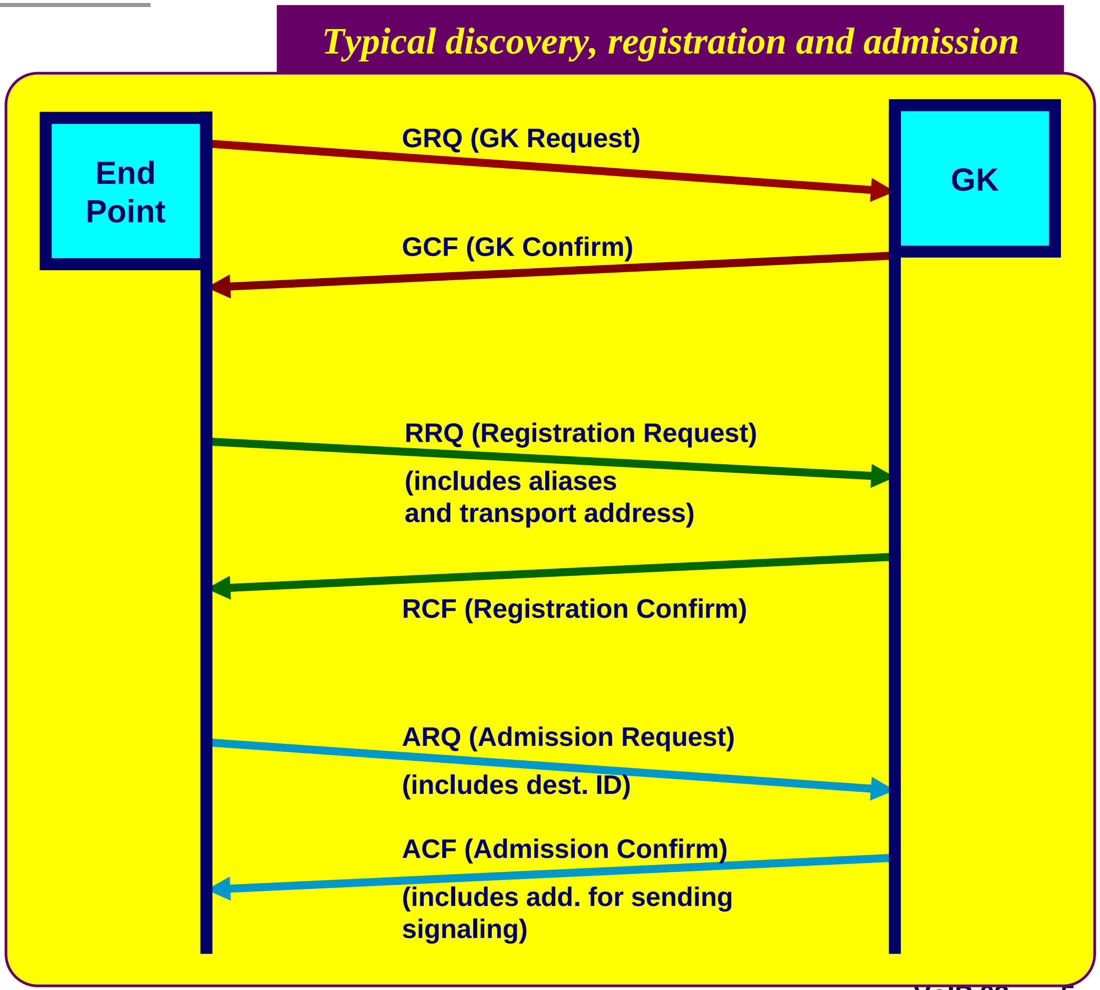
VoIP STANDARDS

HANDSHAKING STANDARDS

- ***ITU-T H.323 suite***

RAS - Registration, Admission & Status





VoIP STANDARDS

HANDSHAKING STANDARDS

- ***SIP suite***

SIP - Session Initiation Protocol (RFC 2543)

SIP RESPONSES

RESPONSE type	code	Selected examples
INFORMATIONAL	1xx	
SUCCESS	2xx	
REDIRECT	3xx	**The call needs more processing before it can be determined whether it can be completed or not** - 300 - The address in the request resolved in more than one choice. The options are returned to caller. - 301 - Called party has moved permanently. The new location is listed in the response. - 302 - Called party has moved temporarily
CLIENT REQUEST FAILURE	4xx	
SERVER FAILURE	5xx	
GLOBAL FAILURE	6xx	