VoIP Reference Card

H.323

H.323 Protocols

H225	signaling
Q931	included in H225. Caller and callee, Release cause
H245	Media negociation
H450	addition to H225 (Message waiting indicator, DTMF...)

H225 messages

SETUP	initiating a conference
ALERTING	other endpoint is aware (phone ringing)
CONNECT	other endpoint has accepted the conference
CALL PROCEEDING	other endpoint needs more time to connect
RELEASE COMPLETE	other endpoint has finished the conference
FACILITY	message to convey many different information
PROGRESS	a media will be played without connection (ringback tone)

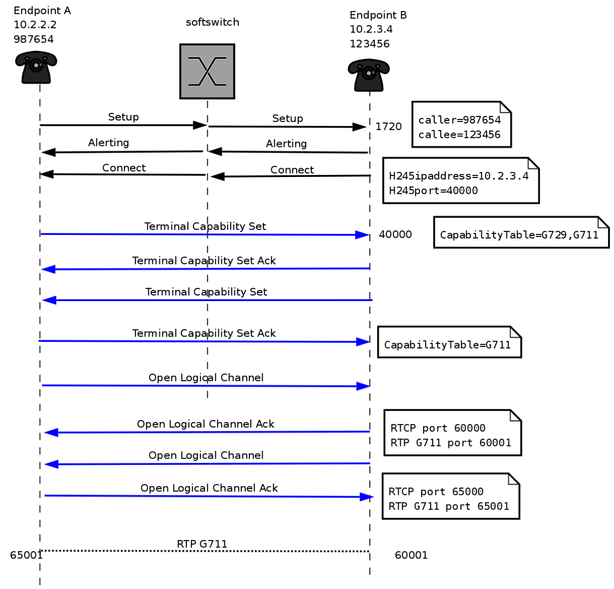
H245 messages

MasterSlaveDetermination	Who is the master? (multi party conference)
TerminalCapabilitySet	The codecs I can receive
OpenLogicalChannel	The address and codec I am listening to
CloseLogicalChannel	close the media stream
EndSessionCommand	close the H245 session

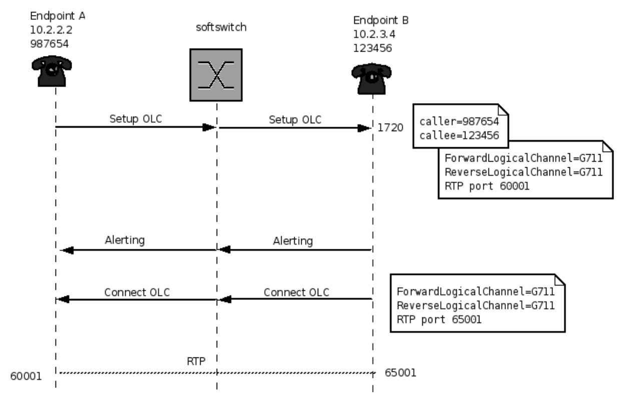
Q.931 Release Causes

1	Unassigned number
3	No route to destination
6	Channel unacceptable
16	Normal call clearing
17	User busy
18	User not responding
19	User alerting; no answer
22	Number changed
27	Destination out of order
28	Invalid number format
34	No circuit/channel available
42	Switching equipment congestion
127	Interworking, unspecified

Simple H.323 Call



FastStart H323 Call



RTP

RTP Common Payloads

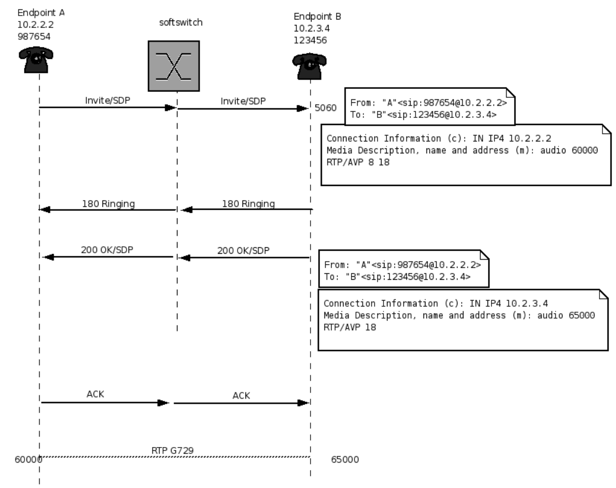
0	PCM, law μ (G.711μ)
8	PCM, law A (G.711A)
9	G.722
4	G.723
15	G.728
18	G.729

SIP

SIP RFC

RFC3261	Session Initiation Protocol
RFC3262	Reliability of Provisional Responses
RFC3263	Locating SIP Servers
RFC3264	An Offer/Answer Model with Session Description Protocol (SDP)
RFC3265	Specific Event Notification
RFC3428	Extension for Instant Messaging
RFC3515	Refer Method (call transfer
RFC2327	Session Description Protocol
RFC3550	RTP

Simple SIP call

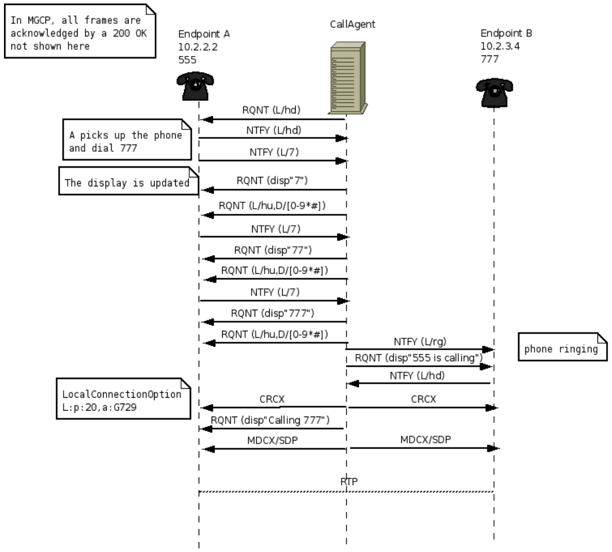


SIP response causes

1xx	Provisional	request received, continuing to process the request
2xx	Success	the action was successfully received, understood, and accepted
3xx	Redirection	further action needs to be taken in order to complete the request
4xx	Client Error	the request contains bad syntax or cannot be fulfilled at this server
5xx	Server Error	the server failed to fulfill an apparently valid request
6xx	Global Failure	the request cannot be fulfilled at any server

MGCP

MGCP Simple Call



MGCP Messages

EPCF	EndpointConfiguration
RQNT	NotificationRequest
NTFY	Notify
CRCX	CreateConnection
MDCX	ModifyConnection
DLCX	DeleteConenction
AUEP	AuditEndPoint
AUCX	AuditConnection
RSIP	RestartInProgress

SS7

Each E1 contains 32 CIC. Usually one or 2 is for signaling and the rest for media.

SS7 Norms

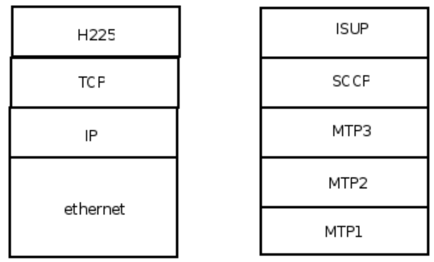
Q.761	Functional Description of ISUP
Q.762	ISUP messages and signals
Q.763	ISUP formats and codes
Q.764	Signaling procedures
Q.767	ISUP for intrnational interdfconnexions
H.264	Interconnection with H323

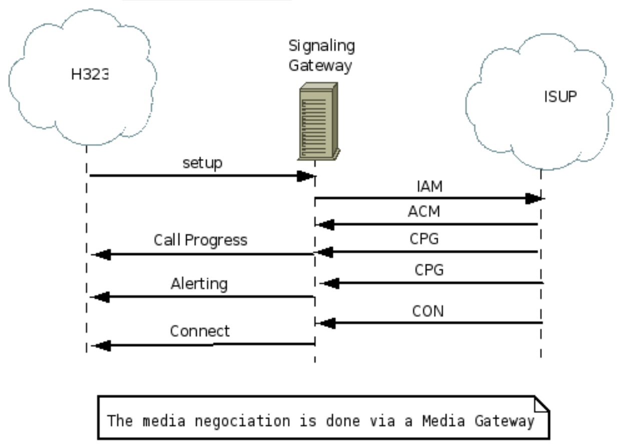
SS7 Common messages

48 different messages defined

IAM	Initial Address Message
ACM	Address Complete
CPG	Call Progress
CON	Connect
REL	Release

H323 Interco





Acronyms

SBC	Session Border Control	Firewall an NAT traversal for VoIP network
ITU	International Telecom Union	
IETF	Internet Engineering Task Force	
SIP	Session Initiation Protocol	Defined by IETF
GK	Gate Keeper	H323 Registration Server
Registrar	like a GK but in SIP	
PBX	Private Branch eXchange	enterprise phone equipment
IVR	Interactive voice response	Vocal application controled by DTMF tones or spoken words
CTI	Computer telephony integration	Control your phone from your computer
3GPP	3rd Generation Partnership Project	Union that defined UMTS & IMS
CPE	Customer Premise Equipment	phone
CLIP	Calling Line Identifier Presentation	Non-anonymous call
CLIR	Calling Line Identifier Restriction	Anonymous call
DTMF	Dual-tone multi-frequency	Dialing a digit during a call

References

voip info

`http://www.voip-info.org/wiki`

List of isup messages

`http://www.acacia-net.com/wwwcla/protocol/ss7_isup.htm`

H323 norms

`http://www.itu.int/rec/T-REC-H.323/f`

book

`La voix sur IP, O. Hersent, D. Gurle, J-P. Petit, Dunod`

About

`VoIP Reference card - Version: 0.1 - 2007`
`Last version : http://voip-refcard.tuxfamily.org/`