VOICE TRANSMISION OVER PACKET NETWORKS

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ABSTRACT

This is a short review about voice transmission techniques over packet based networks. This area is popular today, because of the communications costs saving. In this paper a short description of connection elements needed by the integrated PSTN and VoIP based network is described. Network reliability and sound quality are also functions of the voice-encoding techniques and associated voice-processing functions of the gateway servers. A review of the standards needed by VoIP communication is presented.

1 INTRODUCTION

Many dial-around-calling schemes available today already rely on VoIP backbones to transfer voice, passing some of the cost savings to the customer. These high-speed backbones take advantage of the convergence of Internet and voice traffic to form a single managed network. But along with the initial excitement, customers are worried over possible degradation in voice quality. Whether these concerns are based on experience with the early Internet telephony applications, or whether they are based on understanding the nature of packet networks.

2 CONNECTION ELEMENTS

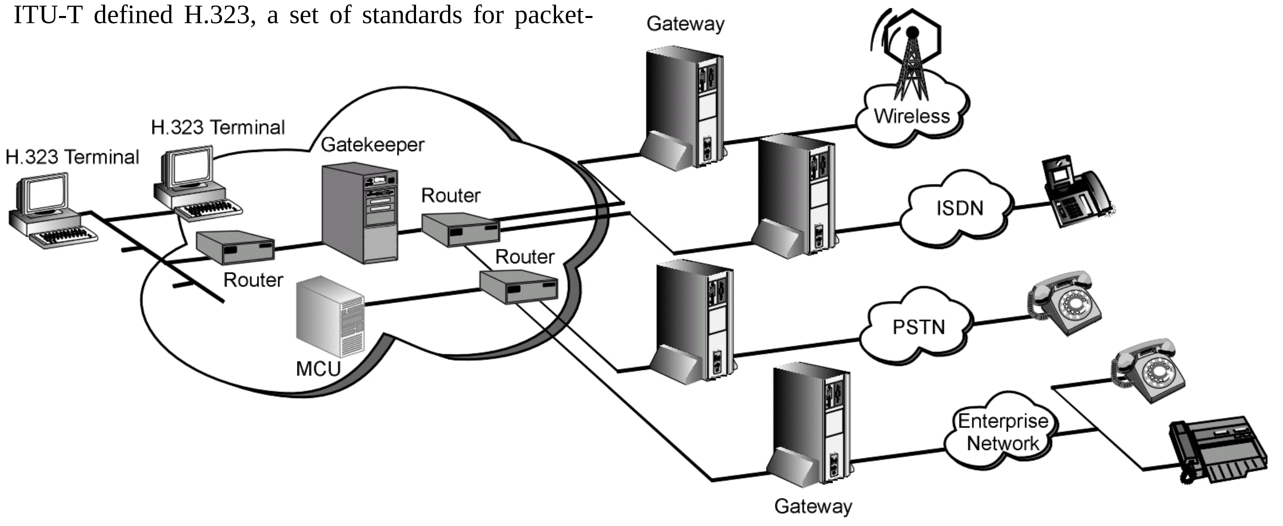
VoIP services need to be able to connect to traditional circuit-switched voice networks. The IETF defined RC2543 - SIP: Session initiation protocol and the ITU-T defined H.323, a set of standards for packet-based multimedia networks. The basic elements of the H.323 network are shown in the diagram below.

H.323 terminals are LAN-based end points for voice transmission. All H.323 terminals support real-time, 2-way communications with other H.323 entities. H.323 terminals implement voice transmission functions and specifically include at least one voice CODEC, signaling functions that are used for call set-up, tear down and so forth. The applicable standards here are H.225.0 signaling which is a subset of ISDN's Q.931 signaling; H.245 which is used to exchange capabilities such as compression standards between H.323 entities; and RAS (Registration, Admission, Status) that connects a terminal to a gatekeeper.

The *gateway serves* as the interface between the H.323 and non-H.323 network. It connects the PSTN or ISDN network to the packet-based devices. The gateway needs to translate signaling messages between the two sides as well as compress and decompress the voice.

The *gatekeeper* is not a mandatory entity in an H.323 network. Gatekeepers manage H.323 zones, logical collection of devices (for example: all H.323 devices within an IP subnet - address translation (routing)). Multiple gatekeepers may be present for load-balancing or hot-swap backup capabilities. Another important function for gatekeepers is providing admission control, specifying what devices can call what numbers.

MCU's allow for conferencing functions between three or more terminals. MCU contains 2 parts.



Multipoint controller (MC) that handles the signaling and control messages necessary to setup and manage conferences. *Multipoint processor* (MP) that accepts streams from endpoints, replicates them and forwards to the correct participating endpoints.

3 AUDIO CODECS

Voice channels occupy 64 Kbps using PCM, but most H.323 devices today use CODECs, that were standardized by standards bodies such as the ITU-T for the sake of interoperability across vendors. Applications such as NetMeeting use the H.245 protocol to negotiate which CODEC to use according to user preferences and the installed CODECs. Different compression schemes can be compared using four parameters: *compressed voice rate*, *complexity* (CPU requirements), *voice quality*, *digitizing delay*.

4 PROTOCOL STACK

Control messages (Q.931 signaling, H.245 capability exchange and the RAS protocol) are carried over the reliable TCP layer. Media traffic is transported over the unreliable UDP layer and includes two protocols as defined in IETF RFC 1889: RTP that carries the actual media and RTCP that includes periodic status and control messages.

Packet size selection is also about balance. Larger packet sizes significantly reduce the overall bandwidth but add to the packetization delay as the sender needs to wait more time to fill up the payload.

Overhead in VoIP communications is quite high. For example 54 bytes overhead to transmit a 20-byte payload. There are two solutions to the problem: *increase packet size*, employ *header compression* (commonly called CRTP or Compressed RTP).

5 PROTOCOL STANDARDS

Signaling

ITU-T Standards and Recommendations

H.323 V2	Packet-based multimedia comm. systems
H.225.0	Call signaling protocols and media stream packetization for packet-based multimedia
H.225.0 AnnexG	Gatekeeper to gatekeeper (inter-domain) c.
H.245	Control protocol for multimedia commun.
H.235	Security & encryption multimed. terminals
H.450.x	Supplementary services for multimedia: 1. Generic functional protocol for the support of supplementary services in H323 2. Call transfer, 3. Diversion, 4. Hold, 5. Park & pickup, 6. Call waiting, 7. Message waiting indication
H.323 Annex D	Real-time fax using T.38
H.323 Annex E	Call connection over UDP
H.323 Annex F	Single-use device
T.38	Proc. for realtime group3 facsimile over IP
T.120 series	Data protocols for multim. conferencing

IETF RFCs and Drafts

RFC 2543	SIP: Session initiation protocol
RFC 2327	SDP: Session description protocol
Internet Draft	SAP: Session announcement protocol

Gateway Control

ITU

H.GCP	Recommendations for gateway control prot.

IETF

Internet Draft	MGCP: Media gateway control protocol
Internet Draft	MEGACO protocol
Draft	SGCP: Simple gateway control protocol
Internet Draft	IPDC: IP device control

Media Transport

IETF

RFC 1889	RTP: Real-time transport protocol
RFC 1889	RTCP: Real-time transport control protocol
RFC 2326	RTSP: Real-time streaming protocol

Media Encoding

ITU

Voice

Standard	Algorithm	(Kbit/s)	(ms)	Voice Q.
G.711	PCM	48, 56, 64	<<1	Excellent
G.723.1	MPE/ACELP	5.3, 6.3	67-97	G(6.3),F(5.3)
H.728	LD-CELP	16	<<2	Good
G.729	CS-ACELP	8	25-35	Good
G.729 a. A	CS-ACELP	8	25-35	Good
G.722	S.B. ADPCM	48, 56, 64	<<2	Good
G.726	ADPCM	16,24,32, 40	60	Good(40), Fair(24)
G.727	AEDPCM	16,24,32, 40	60	Good(40), Fair (24)

Video

Standard	Algorithm	(Kbit/s)	Picture Q.
H.261	DCT with motion compensation	px64 (p=# of ISDN B ch.)	Low
H.263	Improved ver. H261	Various	Medium

6 CONCLUSIONS

Internet telephony promises to combine our separate data and voice networks into a single transport mechanism. However, there are significant barriers in the form of tradeoffs; finding the best combination of codec, access technology, and end-to-end architecture is challenging.

7 REFERENCES

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