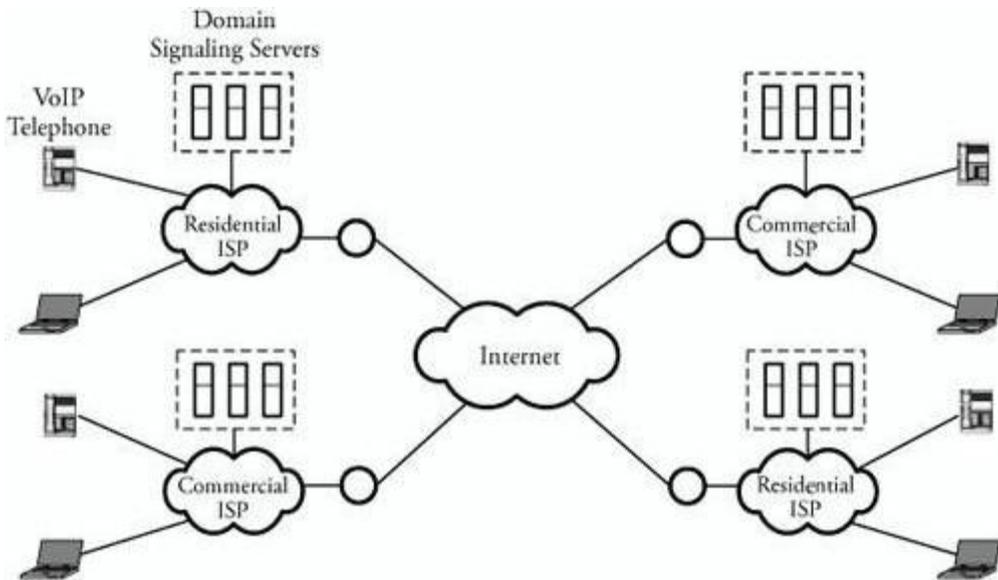


OVERVIEW OF IP TELEPHONY

An IP telephone can be used to make telephone calls over IP networks. Voice over IP (VoIP), or IP telephony, uses packet-switched networks to carry voice traffic in addition to data traffic. The basic scheme of IP telephony starts with pulse code modulation, discussed in [Chapter 7](#). The encoded data is transmitted as packets over packet-switched networks. At a receiver, the data is decoded and converted back to analog form. The packet size must be properly chosen to prevent large delays. The IP telephone system must also be able to handle the signaling function of the call setup, mapping of phone number to IP address, and proper call termination.

Basic components of an IP telephone system include IP telephones, the Internet backbone, and signaling servers, as shown in [Figure 7.1](#). The IP telephone can also be a laptop or a computer with the appropriate software. An IP telephone connects to the Internet through a wired or a wireless medium. The signaling servers in each domain are analogous to the central processing unit in a computer and are responsible for the coordination between IP phones. The hardware devices required to deploy packet-switched networks are less expensive than those required for the connection-oriented public-switched telephone networks. On a VoIP network, network resources are shared between voice and data traffic, resulting in some savings and efficient use of the available network resources.

Figure 7.1. Voice over IP system



A VoIP network is operated through two sets of protocols: signaling protocols and real-time packet-transport protocols. Signaling protocols handle call setups and are controlled by the signaling servers. Once a connection is set, RTP transfers voice data in real-time fashion to destinations. RTP runs over UDP because TCP has a very high overhead. RTP builds some reliability into the UDP scheme and contains a sequence number and a real-time clock value. The sequence number helps RTP recover packets from out-of-order delivery. Two RTP sessions are associated with each phone conversation. Thus, the IP telephone plays a dual role: an RTP sender for outgoing data and an RTP receiver for incoming data.

Source : <http://elearningatria.files.wordpress.com/2013/10/cse-vi-computer-networks-ii-10cs64-notes.pdf>