IP-Telephony

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Abstract— This paper discusses about the new trends in internet protocol telephony and its structural format. Various parameters of this system were explained. It concludes with an overview of the network and its structural aspects.

Key words: Internet protocol, Telephony

I. INTRODUCTION

Voice-over-Internet protocol (VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the internet. The term Internet telephony specifically refers to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the public Internet, rather than via the public switched telephone network. The steps and principles involved in originating VoIP telephone calls are similar to traditional digital telephony and involve signaling, channel setup, digitization of the analog voice signals, and encoding. Instead of being transmitted over a circuit-switched network, however, the digital information is packetized, and transmission occurs as Internet Protocol (IP) packets over a packet-switched network. Such transmission entails careful considerations about resource management different from time (TDM) networks. Early providers of voice-over-IP services offered business models and technical Solutions that mirrored the architecture of the legacy telephone network. Second-generation providers, such as Skype, have built closed networks for private user bases, offering the benefit of free calls and convenience while potentially charging for access to other communication networks, such as the PSTN. This has limited the freedom of users to mix-and-match third-party hardware and software. Third-generation providers, such as Google talk have take on board the concept of federated VoIP—which is a departure from the architecture of the legacy networks. These solutions typically allow dynamic interconnection between users on any two domains on the Internet when a user wishes to place a call. The Internet, and Internet Protocol (IP)-based networks, is increasingly being used as alternatives to the traditional circuit-switched telephone service. The many different 'flavours' of IP Telephony provide, to varying degrees, alternative means of originating, transmitting, and terminating voice and data transmissions which would otherwise be carried by the public switched telephone network (PSTN). In many countries it is now possible, using a standard telephone, to call almost any other telephone in the world by means of IP Telephony, for some or the entire route travelled by the call. These calls are mainly carried outside of the PSTN, and hence outside the regulatory and financial structures which have grown up around it. Originally a curiosity among computer hobbyists possessing similarly-equipped personal computers (PCs), IP Telephony technology now represents a fully-fledged alternative to traditional circuit-switched telecommunication equipment and services. . While IP-based networks are optimized for the carriage of data rather than voice, they can nevertheless carry voice very competently and cheaply. In this paper, "IP Telephony" is used as a generic term for the many different ways of transmitting voice, fax and related services over packet-switched IP-based networks. IP Telephony can be subdivided into two major groups: Internet Telephony and Voice-over-IP (VoIP), the difference being the nature of the underlying IP network. Even within these two broad groups, there is a potentially infinite number of ways to use IP technology to provide different services in different ways.

II. TECHNICAL ASPECTS

A. Packet Switching:

The most important concept in IP Telephony technology is packet switching. However, the significance of the difference between packet switching and circuit switching is probably not as great as initially proclaimed by the Internet community. As IP Telephony development has preceded, the primary goal has been to replicate the functionality, reliability, and service quality of circuit-switched telecommunications networks, and to do so using a protocol suite which was not optimized to facilitate real- time communications. While Internet communication is indeed "connectionless" (that is, no unique end-to- end circuit is created and held for the duration of each call), current development trends seek to make IP Telephony much more connection-oriented than other types of IP communications.

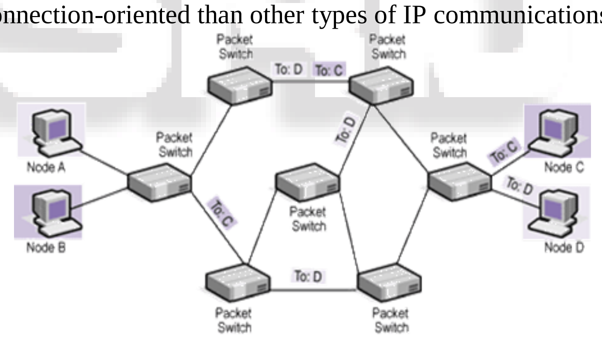

Fig. 1: Packet Switching

B. Gateways:

The link between an IP network and a telephone network is called a Gateway. This is the point at which voice signals become digitized and packetized, or at which digitized packets are converted back to voice. In the case of Phone-to-Phone IP Telephony (including calling card), the gateway is a server which customers dial, much as they might dial an ISP's server from their computer modem. The originating server prompts the user to input an account code and the desired telephone number, and then encodes the ensuing call into IP packets, each with a header directing it to another gateway server somewhere else, where the process will be reversed, and the IP call forwarded to an ordinary telephone. On the other end, the terminating server, which is located as close as possible to the called party's exchange, dials the called party's telephone number and, when a connection is made, starts sending the caller's speech and transmitting the called party's speech in the other direction. Gateways thus allow long distance or international calls to 'look' to the billing systems of PSTN operators at either end like local calls. Once again, it is important to note that not all originating gateways redirect PSTN calls onto the Internet, and not all terminating gateways receive calls from the Internet.

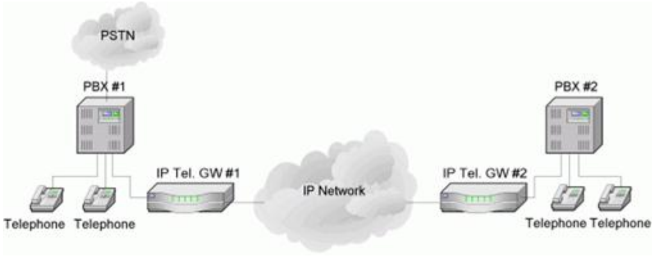

Fig. 2: IP-Telephony Gateway

Gateways can be connected to any kind of IP network, and, in the case of commercial IPTSPs like DeltaThree.com, the network in question is what the company calls "the world's largest network dedicated entirely to the transport of Voice over Internet Protocol.

C. Band width:

The other basic means of decreasing latency in IP packet transmission is to increase the bandwidth of the network or networks employed. More bandwidth means less congestion, which in turn means less delay, and more natural voice conversations. Indeed, some observers argue that increasing available bandwidth is a far more practical means of speeding up the Internet than QoS, because it does not require coordinated action across the Internet.15 However, the Internet community is still trying to figure out how the providers of increased bandwidth, especially for transit, will be compensated for their contributions to overall Internet performance.

III. PROTOCOLS

Voice over IP has been implemented in various ways using both proprietary protocols and protocols based on open standards. Examples of the VoIP protocols are:

- H.323
- Media Gateway Control Protocol (MGCP)
- Session Initiation Protocol (SIP)
- H.248 (also known as Media Gateway Control (Megaco))
- Real-time Transport Protocol (RTP)
- Real-time Transport Control Protocol (RTCP)
- Secure Real-time Transport Protocol (SRTP)
- Session Description Protocol (SDP)
- Inter-Asterisk eXchange (IAX)
- Jingle XMPP VoIP extensions
- Skype protocol
- Teamspeak

The H.323 protocol was one of the first VoIP protocols that found widespread implementation for long-distance traffic, as well as local area network services. However, since the development of newer, less complex protocols such as MGCP and SIP, H.323 deployments are increasingly limited to carrying existing long-haul network traffic. In particular, the Session Initiation Protocol (SIP) has gained widespread VoIP market penetration.

These protocols can be used by special-purpose software, such as Jitsi, or integrated into a web page (web-based VoIP), like Google Talk.

IV. CONCLUSION

The major impact of IP Telephony on Public Telecommunication Operators is likely to be loss of income from international calling, both direct (loss of collection charges) and indirect (loss of settlement payments). But arguably this would happen even without IP Telephony. Markets for international calling are shrinking in value as, on the one hand, prices fall precipitously, while, on the other hand, traffic is routed on least cost routes and settlement rates are forced closer to costs. Operators in developing countries may be better advised to embrace IP Telephony, and bear the consequences of reduced per-minute revenues from long-distance and international services, than to risk missing the opportunity to develop revenues in future growth areas.

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