IP Telephony

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Abstract-

This research paper illustrates the use of IP i.e. Internet Protocol as a medium for voice conferencing, video conferencing and any other media conferencing. One can do audio chat via internet in an efficient manner. It will be easier for the individual on internet network for audio/video conferencing with this cost efficient technique which will only an internet connection. The quality of conferencing will be enhanced by the method of IP telephony.

Keywords: IP, ISP, VOIP

Introduction

IP is an abbreviation for Internet Protocol. It is a communications protocol developed to support a packet-switched network. The protocol has been developed by the Internet Engineering Task Force (IETF). IP telephony is the exchange of information primarily in the form of speech that utilizes a mechanism known as Internet Protocol. IP TELEPHONY is a technique or a communication protocol which is used to transmit the voice and multimedia sessions over internet protocol such as internet. It can also be known as voice over IP broadband phone. IP telephony although does not have globally spread but its expanding rapidly day by day. Voice is transmitting over by integrating internet protocols. VOIP is transmitting of voice by using many different techniques such as media channel setup, digitization of the analog voice signals, encoding packetisation and packet-switched network.

The possibility of transmitting voice over IP-based networks, with all the challenges and associated opportunities, such as voice and data integration, constitutes a milestone in the convergence of the ICT sector. The "IP telephony" topic has been taboo for both its supporters and its opponents, two camps that are highly divided. After due consultation with the Directors of the ITU Telecommunication Standardization Bureau and the ITU Radio communication Bureau, BDT undertook to put the issue on the table for discussion in accordance with the terms of Opinion D Part 3 adopted by the third World Telecommunication Policy Forum (Geneva, 7-9 March 2001).

Introduction to the different types of IP telephony

According to the nature of the IP network used, we may speak of two major categories for voice transmission over IP networks. The first is essentially based on the Internet, which is seen as the interconnection of a host of public or private networks on a global scale. The second is provided by service operators using managed IP networks, within which a number of pre-installed mechanisms (routing algorithms, coding, etc.) serve to ensure a quality of service level that is acceptable for speech. There are three Voice over Internet Protocol (VoIP) usage scenarios according to terminal equipment and types of network:

Scenario 1: PC to PC

In this scenario, the calling and called parties both have computers1 that enable them to connect to the Internet, usually via the network of an Internet service provider (ISP). The two correspondents are able to establish voice communication only by prior arrangement, since both users have to be connected to the Internet at the same time (having fixed in advance the time at which they will communicate via the Internet, unless of course they are permanently online) and use VoIP-compatible software3. Furthermore, the caller must know the IP address of the called party; to overcome this, correspondents must agree to consult an online directory server (updated with each connection) where users register prior to each communication or have other ways of locating or being aware of the availability of their correspondent's connection to the Internet (Instant Messaging technologies).

Scenario 2: Phone-to-phone over IP

In this case, the calling and called parties are both subscribers to the public telephony network (fixed or mobile) and use their telephone set for voice communication in the normal way. There are two methods for communicating by means of two ordinary telephone sets via an IP or Internet network.

Use of gateways

This means that one or more telecommunication players have established gateways that enable the transmission of voice over an IP network in a way that is transparent to telephone users. What we have in this case is not the Internet but a "managed" IP network, i.e. a network which has been dimensioned in such a way as to enable voice to be carried with an acceptable quality of service. Figure 2 below illustrates such a scenario.

Scenario 3: PC-to-Phone or Phone-to-PC

In this scenario, one of the users has a computer by which he connects to the Internet via an access network and an ISP (in a similar way to scenario 1)5, while the other user is a "normal" subscriber to a fixed or mobile telephone network.

PC-to-Phone

When the computerized user wishes to call a correspondent on the latter's telephone set, he must begin by connecting to the Internet in the traditional manner via the network of his ISP. Once connected, he uses the services of an Internet telephony service provider (ITSP) operating a gateway which ensures access to the point that is closest to the telephone exchange of the called subscriber. It is this gateway that will handle the calling party's call and all of the signalling relating to the telephone call at the called party end. It should be noted that the ITSP provides a one-way PC-to-phone service and does not manage subscribers as such; in fact, the PC subscriber uses the ITSP's services solely for outgoing calls. It should also be noted that the ITSP has a managed IP network, thereby ensuring a certain quality of service for voice as far as the gateway closest to the called subscriber, and that the ITSP also manages the interconnection with the latter's telephone operator. Despite their use of VoIP technology, ITSPs consider themselves to be telephone service providers and generally provide their services to individuals in the conventional manner, i.e. with a charge per minute.

Phone-to-PC

In this case, the calling party is the telephony user and the called party is the PC user. Since a telephony user can essentially dial an E.164 number to reach the called party, then somehow the PC user should have an E.164 number:

- Either indirectly: In the case of its interconnection to the network behind an IP-technology Private Branch Exchange (PABX) switch (actually, in this case we can more properly speak about an "IP Phone" rather than a PC device that is connected to the LAN managed by the IP PABX).
- Or directly: In this case, it the IP side subscriber who has an E.164 address allocated by an IP telephony operator.

NETWORK ARCHITECTURE

NETWORK INFRASTRUCTURE

The telecommunication network may be described as a set of infrastructures and complete set tools and integrated technologies for deploying man aging network service that support IP communication. The concept of network arises from the need to share infrastructures in order to improve efficiency cost, it being possible for a single line utilized by different users for different periods. The architecture includes intelligent switches, routers and specialized components which form physical infrastructures to provide the services such as service, quality of service, resiliency and many applications.

LEGACY TELECOMMUNICATION ARCHITECTURE

Telephony networks have gone under major evolutionary changes, driven essentially by technological progress in various fields. The new updated technology now days are digitization of their transport technology. The current technology which we are using “circuit switching”. It is based on the principle that once a circuit has established connection must be reserved from the setting up of a call to its conclusion. The quality-of-service requirement implies that adequate resources (circuit capacities, transmission speeds, and management arrangements) must be mobilized throughout the duration of a call in each of the sub networks implicated in the call between the two communicating parties. This has a bearing not only on the technology used to carry voice, but also, and more fundamentally, on the very design of the logic incorporated in the network's active components (switches) and of the mutual language (signalling) they use to ensure the proper routing of a call between two or more subscribers.

DATA NETWORK ARCHITECTURE

Data networks have been designed initially to interconnect computers their servers and management platforms. This architecture mainly used for the transfer of data between machines in computer applications. In this the application they support are generally transmit by using packet switched technology. This flexibility had led, for instance, to the use of data networks (especially IPhones) for new types of "human-related" communication applications, like voice and video Transmission, and through – still negligible but likely to take off in the coming years – appliance Devices capable of executing a given subset of communication applications without the need of having a "general-purpose" computer device.

TELECOMMUNICATION NETWORK

The interconnection of personnel computers to data network become apparent, the most valid choice to access the network become global telecommunication network.

The reason for choosing this network is-

1. The presence of network everywhere at the same time, particularly in developed areas.

2. The availability of modems for converting digital information from the personal computer into analog signals for the transmission over the telephone network.

APPLICATION

Benefits to end users

In addition to providing the potential for lower cost telephony to end users, IP telephony technology makes it easier to create new applications capabilities because it:

• carries and processes voice, data and multimedia traffic and signalling in the same form;

• utilizes the Internet Protocol and the host of open, standardized interfaces and software languages available to it.

• Examples of such capabilities and applications are: • IP Centrex – extends traditional Centrex capabilities to not only accommodate voice, but also data and multimedia;

• Unified Messaging – delivers voice, fax and e-mail messages into a single mailbox that users can access anywhere from a web browser, e-mail, or a telephone;

• Pre/Post-paid calling – offers a range of pre- and post-paid calling card capabilities created on an open platform;

• Internet call waiting – enables a single phone line for both voice calls and Internet access;

• Conference call capabilities;

• Call/Contact centres – enables a range of call centre capabilities, e.g. web enabled.

VoIP virtual trunking

In this application, the IP network replaces the TDM trunk network. Calls originating at the PSTN are passed to the IP network at a gateway, which also converts the media stream, carried over the IP network over a pre-provisioned virtual trunk (e.g. layer 2 tunnel) to a gateway at the terminating PSTN, where the media stream is converted back and delivered to the called party. Signalling between the PSTNs uses BICC carried over SCTP.

At present, VoIP, when provided as a public service, is supposed to use E.164-based addressing (in the private trunking application it may follow a private numbering plan), whereas Internet telephony uses web-based addressing. With the progress of work in ENUM (see Annex H), this may change in the future.

Multimedia applications

Delivery of multimedia information and multimedia communication in a unified way is considered as one driver for creating the IPTN+, as it allows the creation of new applications/ services to become a new source of revenue.

CONCLUSION

IP telephony service providers include or soon will include local telephone companies' long distance providers such as AT&T, cable TV companies, ISPs, and fixed service wireless operator. Currently, unlike traditional phone service, IP telephony service is relatively unregulated by the government. In the United States, the Federal Communications Commission (FCC) regulates phone-to-phone connections, but says they do not plan to regulate connections between a phone user and an IP telephony service provider. VOIP is an organized effort to standardize IP telephony. IP telephony is an important part of the convergence of computers, telephones, and television into a single integrated information environment

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