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# A COST EFFECTIVE VOICE NETWORK USING THE VoIP TECHNOLOGY

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## *ABSTRACT*

*In this paper, a cost effective voice network using the VoIP (Voice over Internet Protocol) technology is described. The connections of Private Branch Exchanges (PBXs) to a VoIP network, to the PSTN (Public Switched Telephone Network) and to the ISDN (Integrated Services Digital Networks) are considered from the technological point of view; the VoIP protocol architecture, the VoIP standards, quality of service (QoS) issues in the VoIP networks are discussed, traffic load and channel capacity calculations of a sample application are presented. It is emphasized that government organizations, universities, big companies, banks and other establishments, having units or campuses situated in different sites in the same city or in different cites can make use of the VoIP technology to reduce their telephone costs. Although the VoIP technology is ready for such applications, there is an argument in Turkey whether or not VoIP is legal even for intranet applications. It is mentioned that the regulations concerning VoIP in Turkey need to be clarified or rearranged to stop the argument and enable the use of VoIP Technology.*

**Keywords:** *VoIP standards, PBX, quality of service, Erlang, traffic calculations, channel capacity*

## 1. INTRODUCTION

The VoIP technology has changed the traditional application of sending data over voice networks to sending voice over data networks. As covered under the umbrella of ITU-T 323 recommendations, multimedia (voice, data and video) communication can be carried over the Internet Protocol.

Telephone costs of governmental organizations, universities, big companies, banks and other establishments, having units or campuses situated in different sites in the same city or in different cites are generally quite high mainly due to the calls made between different PBXs of the same establishment. In such cases, the VoIP technology can be applied to reduce telephone costs. For the purpose of cost reduction in telephone conversations, applications were started in 1995 at hobby level for voice communication over the Internet by using specially equipped PCs. Infrastructure studies for carrying voice and data together over the Internet were started in 1999 in the developed countries.

Voice communication over the Internet Protocol has created two new concepts: (1) Internet

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Telephony or IP telephony, (2) Voice over Internet Protocol (VoIP). Actually, the first one includes the second one. In both types of these communications, analog voice signal is first converted into a digital voice signal and subjected to data compression to reduce transmission speed, and then placed in IP (Internet Protocols) packets to be sent over the Internet or in the VoIP network. In the Internet telephony (or IP telephony), the users transfer audio data to and from PSTN over suitable gateways, over a WAN (Wide Area Network) or the Internet by using the Internet protocol to establish a telephone call to a telephone subscriber in anywhere in the world. The building blocks of IP telephony are presented in [1], and Voice Service Interworking for PSTN and IP Networks is presented in [2]. On the other hand, in VoIP networks, the user transfers audio data over an IP network that is usually a LAN or an intranet to connect members of the same organization for internal phone calls.

In this work, a VoIP network is described from the technological point of view. However, the regulations concerning VoIP in Turkey need to be clarified or rearranged to stop the ongoing argument whether or not VoIP is legal even for intranet applications.

## 2. CONNECTION OF PBXs TO A VoIP NETWORK, TO ISDN AND PSTN

In VoIP networks, PBXs of the same organization situated in different sites or campuses are connected to each other over routers and modems to form an intranet. One of the PBXs, generally the one with the highest subscriber capacity, is chosen as the central PBX and all the connections coming from other PBXs brought together in the central PBX to form an Ethernet LAN (Local Area Network in bus topology using the IEEE 802.3 standard).

Connections between the PBXs can technologically be made by using leased lines or by using the frame relay (FR) network or the ATM (Asynchronous Transfer Mode) network. Frame relay connections are cheaper than leased lines for long distances. ATM is generally used for broadband multimedia applications. If QoS is guaranteed in the VoIP network, it can be said that ATM is not advantageous for transmitting only audio data. Therefore, leased lines can be preferred for the connections between the PBXs situate in the same city.

Analog and digital telephone sets of the internal subscribers of the PBXs are connected to the related subscriber modules of those PBXs; the VoIP telephone sets can directly be connected to the VoIP network over the hubs. ISDN-PBXs can be connected to PSTN over analog external lines and to ISDN over digital external lines (ISDN - PRI: Integrated Services Digital Networks - Primary Rate Interface). Each ISDN-PRI connection includes 30 digital communication channels (30 B channels) each with 64 kbps transmission capacity and one signaling channel (D channel) with 64 kbps transmission capacity. The connections between PBXs to the VoIP network, to the ISDN and PSTN are shown in Figure 1.

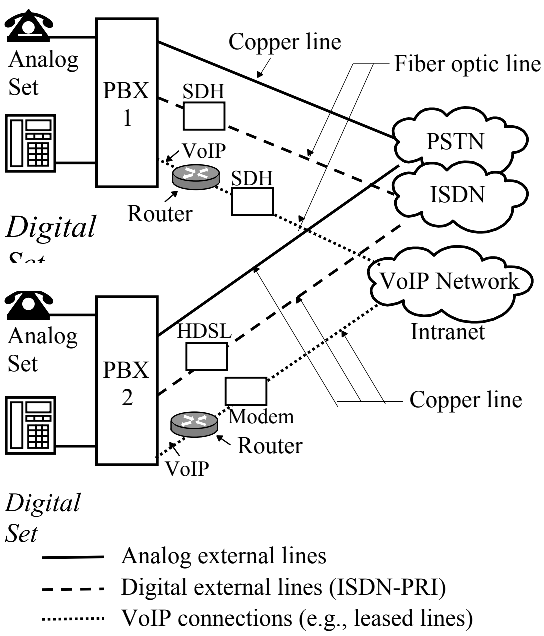


**Figure 1.** Connections of PBXs to the PSTN, ISDN and the VoIP network.

PBXs with a few (for example, less than four) digital external lines can be connected to ISDN over HDSL (High bit rate Digital Subscriber Line) modems and copper lines. PBXs with low VoIP traffic can also be connected to the VoIP network over proper modems and copper lines. However, for the connections between PBXs with more than a few (for example, more than four) digital external lines and ISDN, and also for the connections between PBXs with high VoIP traffic (for example the central PBX) and the VoIP network SDH (Synchronous Digital Hierarchy) equipment and fiber optic lines can be preferred.

For the telephone connections over the VoIP network, the subscribers of the PBXs connected to the VoIP can have all the subscriber features that PSTN subscribers may have. On the digital external lines, DID (Direct Inward Dialing) facility can be used for the incoming calls to the PBXs to make direct connection to the internal subscribers of the PBXs. Calling subscriber's telephone number (or calling subscriber's name, if it is stored in the memory of the called subscriber's telephone set) can be displayed on the display of the digital telephone set of the called subscriber as it is possible in GSM telephones and ISDN telephones.

Voice quality in the VoIP network is controlled by continuously observing the quality of service parameters. When the QoS parameters are detected to be below the threshold level, the overflow traffic between the PBXs is either directed to PSTN or ISDN to establish the connection, or an overflow tone is sent to the subscriber who has made the call attempt.

Benefits of VoIP networks can be stated as follows: (1) VoIP networks provide low cost telephone calls between PBXs situated in different sites since they are private networks and the required voice channel bandwidth is generally four times less than that of PSTN or ISDN. (2) VoIP networks allow unified transmission of voice and data. (3) VoIP networks provide simplicity and flexibility for the centralized network management since both voice and data are transmitted over IP.

## 3. PROTOCOL ARCHITECTURE AND STANDARDS OF VoIP

VoIP protocol architecture is defined by the ITU-T H.323 standard, which is an umbrella recommendation for multimedia communication over LANs [2], [3]. The H.323 standard defines data compression and call control protocols for real time multimedia communication over IP. Figure 2 shows the voice applications related part of the VoIP protocol stack (digitization and compression of voice and inserting audio data into IP packets) of an Ethernet LAN based VoIP network compared with the TCP/IP reference model defined in [4].

| TCP/IP Reference Model | VoIP Protocol Stack |
|---|---|
| Application Layer | Voice Applications |
| | Audio Codecs G.711; G.723.1 G.726; G.727; G.728; G.729A |
| Transport Layer | Real Time Protocol (RTP) |
| | User Datagram Protocol (UDP) |
| Internet Layer | Internet Protocol (IP) |
| Network Access Layer | IEEE 802.3 |
| Physical Layer | IEEE 802.3 |

**Figure 2**. Voice applications related part of the VoIP protocol stack and the TCP/IP reference model.

ITU-T standards pertaining to digitization and compression of voice are given below with their transmission rates, end-to-end delays (excluding the channel delay) and voice qualities indicated.

G.711 PCM (Pulse Code Modulation): 48, 56, 64 kbps; delay << 1 msec; voice quality: Excellent.

G.723.1 MPE/ACELP (Multi-Pulse Excitation Algebraic Codebook Excited Linear Prediction): 5.3, 6.3 kbps; delay: 67-97 msec; voice quality: Good for 6.3 kbps, fair for 5.3 kbps.

G.726 ADPCM (Adaptive Differential PCM): 16, 24 32, 40 kbps; delay: 60 msec; voice quality: Good for 40 kbps, fair for 24 kbps.

G.727 AEDPCM (Adaptive Embedded Differential PCM): 16, 24 32, 40 kbps; delay: 60 msec; voice quality: Good for 40 kbps, fair for 24 kbps.

G.728 LD-CELP (Low Delay Codebook Excited Linear Prediction): 16 kbps; delay << 2 msec; voice quality: Good.

G.729A CS-ACELP (Conjugate Structure-ACELP Annex-A: Reduced Complexity Algorithm): 8 kbps; delay: 25-35 msec; voice quality: Good.

In order to reduce the bandwidth requirement of the VoIP channels, small packets are sent during the silence periods of the speech. This process, which increases the compression efficiency, is called silence suppression. Applying compression on the headers of the IP, UDP and RTP protocols further reduces the bandwidth requirement. This process is called header compression.

The Real-time Transport Protocol (RTP) provides end-to-end network transport functions for applications transmitting real-time data such as audio, video or simulation data over multicast or unicast network services. RTP does not make resource reservation and does not guarantee quality of service for real time services. The RTP Control Protocol (RTCP) works together with the RTP and it controls the reception quality of all users in the session by periodically sending them control packets. If traffic congestion is about to start, the application can reduce the transmission rate.

In real time audio and video communication, retransmission of the lost packets (those are not delivered to the destination) creates more serious problems than not being delivered at all. Therefore, a transport protocol that does not retransmit the lost packets must be chosen. For this reason, in real time audio and video communication, User Datagram Protocol (UDP) is used instead of Transmission Control Protocol (TCP).

## 4. QUALITY OF SERVICE IN VoIP

The advantages of reduced cost and bandwidth savings of sending voice over IP networks are associated with some quality of service issues such as delay, jitter, echo cancellation, silence suppression and lost packet interpolation.

Three types of delay can be considered in VoIP applications: (1) Algorithmic delay that is created in the codecs, (2) Processing delay that is created in processing the packets and (3) Network delay that is caused by the physical medium and the protocols used to transmit the voice data, and by the buffers used on the receiver side to remove packet jitter.

In telephone networks, echo is caused by signal reflections generated by the hybrid circuit that converts between a 4-wire circuit (a separate transmit and receive pair) and a 2-wire circuit (a single transmit and receive pair). The speaker hears the reflection of his own voice. However, it is acceptable when the round trip delay is less than 50 msec since the echo is masked by the normal side tone every telephone generates. Echo becomes problem in VoIP networks when the round trip delay through the network exceeds 50 msec. Echo cancellation techniques are used to solve this problem. The ITU-T G.165 recommendation defines the performance requirements that are required for echo cancellers.

The variation (i.e., standard deviation) in packet arrival times is called jitter. Jitter can be removed by using buffers, which collects packets and holds them long enough to allow the slowest packet to arrive in time to be played in the correct sequence. This causes additional delay. A balance must be established between the packet delay created in the buffer and the total delay.

Lost packets can be a serious problem depending on the packet network used. Lost packet compensation (or lost packet interpolation) methods are used to reduce the effects of this problem. If the lost packet ratio is less than % 10, then retransmission of the last packet instead of the lost packet can prevent degradation of voice quality.

In the VoIP application shown in Figure 1, the QoS parameters are continuously observed and new call attempts are rejected when one of these parameters is detected to be below the threshold level. When a new call attempt is rejected, either an overflow tone is sent to the rejected subscriber or the call is automatically establishment over an analog external line and PSTN or over a digital external line (ISDN-PRI) and ISDN.

## 5. TRAFFIC LOAD AND CHANNEL CAPACITY CALCULATIONS IN VoIP NETWORKS

If we assume that telephone calls arrive as a stationary Poisson process, the traffic-performance relation can be defined by the Erlang loss formula [5] which gives the probability of call blocking or call congestion probability (P) when a certain volume of traffic ($a$) is offered to a given number of voice circuits (n). This formula, which is also known as the Erlang B formula, is given in Equation (1) under the following assumptions:

- Call attempts are made at random instances,
- The number of call sources are infinite,
- The non served calls are lost calls,

- The non served calls do not reattempt,
- All the circuits are authorized.
- The call blocking probability is independent of the conversation duration.

$$P = \frac{a^n / n!}{\sum_{i=0}^{i=n} a^i / i!} \qquad (1)$$

By using Equation (1), any one of the parameters $a$ , n and P can be calculated by using the online calculator given in [6], when the other two is known. Here, the traffic load $a$ defines one Busy Hour Traffic (BTH) load. P takes values between 0 and 1. In practice, P is generally chosen as 0.01. This value corresponds to the case where one call attempt is unsuccessful (or blocked) out of 100 call attempts. In some applications P can be chosen greater than 0.01.

Erlang is a unit of telecommunications traffic measurement. It is used to describe total traffic volume in one hour. The number of Erlangs is equal to the number of calls in one hour multiplied by the average call duration in terms of hours. For example, if a group of users makes 50 calls in one hour, and each call has an average call duration of 3 minutes (3/60 hours), then the number of Erlangs is calculated as follows:

$$a = [\text{Number of calls in one hour}] \times [\text{Average call duration (in terms of hours)}]$$

$$a = 50 \times (3 / 60) = 2.5 \text{ Erlang} \qquad (2)$$

If traffic load is calculated for the busy hour of the day where the traffic load is at the highest value, then the Busy Hour Traffic (BHT) load is obtained.

Traffic and channel capacity calculations of the VoIP network shown in Figure 1 can be explained by an example. Considering a VoIP network consisting of 6 PBXs, the traffic load ( $a$ in terms of Erlangs) at the VoIP connection of each PBX can be calculated by using Equation (3).

$$a = MN_1N_2 / (N_1 + N_2) \qquad (3)$$

Here, $N_1$ is the number of internal subscribers of the PBX of which the traffic calculation is being done; $N_2$ is the total number internal subscribers of the PBXs connected to the VoIP; M is the utilization ratio, and it can be chosen between 0.01 and 0.1 depending on the call characteristics of the internal subscribers of the PBX. Table 1 shows the traffic loads of each PBX that are calculated by using Equation (3) for M=0.05, number of VoIP channels calculated by using and the Erlang B calculator in [6] that is based on Equation (1) for P=0.01, and the VoIP channel capacity by taking transmission rate of each voice channel as 16 kbps.

**Table 1:** The traffic loads, number of VoIP channels and VoIP channel capacities of the PBXs.

| PBX Nr. | $N_1$ | $N_2$ | Traffic (Erlang) | VoIP Channels Quantity | VoIP Channels Capacity (kbps) |
|---|---|---|---|---|---|
| 1 | 2000 | 3150 | 61.17 | 76 | 1216 |
| 2 | 1500 | 3650 | 53.16 | 67 | 1072 |
| 3 | 1000 | 4150 | 40.29 | 53 | 848 |
| 4 | 500 | 4650 | 22.57 | 33 | 528 |
| 5 | 100 | 5050 | 4.9 | 11 | 176 |
| 6 | 50 | 5100 | 2.48 | 7 | 112 |

# 6. CONCLUSION

The VoIP technology provides efficient use the transmission capacity in comparison to PSTN and ISDN for voice communication. The capacity used for a voice channel in PSTN and ISDN is 64 kbps, where as in VoIP networks, generally, 16 kbps channel capacity is sufficient for a voice channel. Another important advantage of VoIP networks is the unified transmission of audio, video and data over the same network, which also provides simplicity and flexibility in network management. In VoIP networks, it is possible to continuously observe the QoS parameters, to give an overflow tone to the caller or to automatically direct the new call over PSTN or ISDN and when one of the QoS parameters falls below the threshold level.

Transition to VoIP will result in cost reduction of telephone calls of big organizations and companies those who have units spread out in various places in the same city or in different cities. However, there is an ongoing argument in Turkey whether a VoIP network built as an intranet is legal or not. Therefore, the laws and regulations pertaining to VoIP applications in Turkey must be clarified or rearranged to stop this argument and enable VoIP applications in Turkey.

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