Implementing VoIP over Fatima Jinnah Women University

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Abstract

Working people employee the use of technology in such very effective natural ways that permit them to do what they want: they can communicate with anyone as per need, in the time and space that suits them the best. Easily accessible and user-friendly, collaboration tools allow them to explore, share, engage, and connect with people and content in meaningful ways that help them learn. Traditional telephony carriers use circuit switching for carrying voice traffic. Circuit switching was designed for voice from the outset; hence it carries voice in an efficient manner. However it is an expensive solution. Nowadays people want to talk much more on phone, but they also want to communicate in a myriad of other ways – through e-mail, instant messaging, video, the World Wide Web, etc. Circuit switching is not suitable for this new world of multimedia communication. Therefore, in this paper a solution is proposed to implement scalable VOIP system over Fatima Jinnah University that will lead to better Quality of service and facilitates enhanced communication.
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Keywords: *Private branch exchange, Real time protocol, Session Initiation Protocol, Voice over Internet Protocol, Internet protocol, Public switched telephone networks, pulse code modulation.*

1. Introduction

In this world of fast growing technological developments and the multimedia communications, need for fast paced, integrated services are required. Whereby the Internet protocol provides a healthy environment for integrated services combining voice and data. VoIP is a well suited solution for current networks such as LAN based corporate networks that replaces relatively expensive and circuit switched based PSTN network. This research paper deals with implementing VoIP over Fatima Jinnah Women University Network. The idea of using a IP-based PBX is quite useful in such a case. Firstly, this system integrates the corporate telephone system with the corporate computer network, removing the need for two separate networks. The PBX itself becomes just another server or group of servers in the corporate LAN, which helps to facilitate voice/data integration.

The proposed system has been developed by implementing a soft PBX which acts as the server which performs call-routing functions, replacing the traditional legacy PBX or key system. This PBX would allow a number of attached soft phones to make calls to one another and to connect to other telephone services. The basic software would include many features available in proprietary PBX systems: voice mail, conference calling, interactive voice response, and automatic call distribution, just to name a few. The proposed system also helps the user in building the dial plan for the network.

The scope of this research project is to facilitate faculty/users and enable new forms of communication and engagement in the classroom, permitting extensions and variations of the informal interactions already occurring in classrooms and hallways, and creating new frontiers for collaboration across geographic boundaries.

2. **Background:**

Since 1990's phone has been a preferred and convenient way to communicate. In the developed world the phone network has grown and now it is almost everywhere. As we have experienced that internet is surely a public network and therefore connection to it is very cheap, merely costing a local call at the most which is totally unlike phone network.

The very first way for communication through VoIP was by running software on a PC. The first internet phone software was released by Vocaltec, Inc. in 1995 [1], this software was designed to be able to run on a 33MHz 486 PC. The use of soft phones on the computer systems is one option for transferring voice over the IP network. On the other hand using phone adapters [2] is another way to enable VoIP services.

VoIP is a technology used to transmit voice conversation over a network using the Internet Protocol [3]. VoIP has nowadays become the hottest telecommunication research topic and as one of them it has many advantages over the traditional Public Switched Telephone Networks (PSTNs), which are discussed:

- Voice/Data Integration and Advanced Services [4]
- Lower Equipment Cost
- Lower Bandwidth Requirements
- Widespread availability of IP.

The protocols and services used in VoIP are detailed in Table 1.

Table 1: VoIP Protocols/Services

Application Layer	Audio RTP, RTCP, (SIP, H.323..)
Transport Layer	UDP
Network Layer	IP
Physical Layer	Ethernet, SDH,…

2.1 RTP

The RTP is an essential streaming protocol. It is evidently essential for real time applications that are designed for synchronization of traffic streams compensating for delay variations and de-sequencing. However not ensure on-time delivery or traffic signals or address the issue of QoS, relevant to guaranteed bandwidth availability for specific applications. RTP is generally used in conjunction with UDP, but it can surely make use of any packet-based lower layer protocol [5].

2.2 RTCP

The RTCP is a companion protocol of RTP that is used to transmit periodic control packets on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets [7]. RTCP provides QoS feedback and session information. The packets of RTCP carry information related to the state of end points and the statistical information regarding the current RTP stream such as delay times, packet loss and jitter [5]. Both RTP and RTCP run on the top of User Datagram Protocol (UDP) that provide better real-time responsiveness and lower overhead. After compression and digitization of a sample of analogue voice signal, RTP and RTCP information can be packed inside an IP packet with an UDP header to accomplish the requests of transmitting a voice packet.

2.3 SIP

The SIP is a signaling protocol which is simple and light weighted, it is most commonly implemented on top of the User Datagram Protocol (UDP), but it can also be implemented on top of the Transmission Control Protocol (TCP) [8].

3. Related Work:

[8] Regards VoIP as appropriate technology for roaming academics and local deployment for telephony services. [9] Claims of the high merits of SIP protocol being used for implementing VOIP that has lead Nokia to earn more business. [10] Research survey carried on 280 companies revealing the cost optimization he driving force for deploying (VoIP Ezilon.com, 2010). [11] discusses the comparative analysis of POTS with VoIP where voice converted to digitized form(binary 0s and 1s) and transferred as packets. [12] The project deployed on a high bandwidth LAN so therefore the efficiencies of the protocols used in VoIP networks are thus not particularly an issue: the range and flexibility of providing services was more important with the use of high quality codec.

4. Research Goal:

The aim of the research paper was to develop an efficient, cost effective and bandwidth optimized soft PBX solution that implements VoIP aimed to facilitate dynamic academics as well as local deployment.

5. Implementation:

The proposed system is implemented as a soft PBX that acts as a server responsible for performing routing functionalities replacing the traditional legacy PBX or key system. The system has been developed in fast growing cutting edge technology of .Net and C# framework. Operating system is windows 2000 or windows XP. The client/server architecture is followed with Soft PBX as server as in Fig 2 and the soft phones acting as clients as in Fig 3. G.729 codec together with SIP and RTP/RTCP protocol is implemented as shown in Fig 1. G.729 codec has been chosen due to variety of factors such as lower bit rates that works efficiently well in congestion conditions. Also adaptive jitter buffers have been implemented to improve performance when compared with fixed jitter buffers. Furthermore, with Forward Error correction receiver recovers from packet loss without retransmissions but it increases the delay.

Session Initiation protocol has been implemented. Unnecessary ports/services in SIP and H.323 gateways have been disabled to decrease the possibility of unauthorized access and remote code execution. Packet loss concealment technique has been implemented in order to improve VoIP performance.

Real Time protocol has been implemented to ensure a proper delivery of packets with real time characteristics. Specifically RTCP is being implemented to provide reception quality feedback of RTP data distribution to all the participants/clients in a session.

```
namespace WaveLib
{
publicclassWaveStream : Stream, IDisposable
    {
    privateStream m_Stream;
    privatelong m_DataPos;
    privatelong m_Length;
    privateWaveFormat m_Format;

publicWaveFormat Format
            {
get { return m_Format; }
            }
privatestring ReadChunk(BinaryReader reader)
{
byte[] ch = newbyte[4];
reader.Read(ch, 0, ch.Length);
return System.Text.Encoding.ASCII.GetString(ch);
            }
privatevoid ReadHeader()
{
BinaryReader Reader = newBinaryReader(m_Stream);
if (ReadChunk(Reader) != "RIFF")
thrownewException("Invalid file format");
            Reader.ReadInt32();
 // File length minus first 8 bytes of RIFF description,
we  don't use it

if (ReadChunk(Reader) != "WAVE")

thrownewException("Invalid file format");

if (ReadChunk(Reader) != "fmt ")

thrownewException("Invalid file format");

int len = Reader.ReadInt32();
if (len < 16) // bad format chunk length

thrownewException("Invalid file format");
        if (m_Stream != null)
        m_Stream.Close();
        GC.SuppressFinalize(this);
        }

    publicoverridebool CanRead
    {
    get { returntrue; }
    }
    publicoverridebool CanSeek
    {
    get { returntrue; }
```

```
m_Format =newWaveFormat(22050,16, 2);//initialize
to any format
m_Format.wFormatTag = Reader.ReadInt16();
m_Format.nChannels = Reader.ReadInt16();
m_Format.nSamplesPerSec = Reader.ReadInt32();
m_Format.nAvgBytesPerSec = Reader.ReadInt32();
m_Format.nBlockAlign = Reader.ReadInt16();
m_Format.wBitsPerSample = Reader.ReadInt16();
// advance in the stream to skip the wave format block
len -= 16; // minimum format size
while (len > 0)
{Reader.ReadByte();
len--;}
// assume the data chunk is aligned
while(m_Stream.Position < m_Stream.Length &&
ReadChunk(Reader) !="data");

if (m_Stream.Position >= m_Stream.Length)

thrownewException("Invalid file format");

m_Length = Reader.ReadInt32();
m_DataPos = m_Stream.Position;
```

```
}
publicoverridebool CanWrite
{
get { returnfalse; }
}
publicoverridelong Length
{
get { return m_Length; }
}
publicoverridelong Position
{
{ return m_Stream.Position - m_DataPos; }
set { Seek(value, SeekOrigin.Begin); }
    }
publicoverridevoid Close()
{
Dispose();
}
publicoverridevoid Flush()
{
}
publicoverridevoid SetLength(long len)
{
thrownewInvalidOperationException();        }
publicoverridelong Seek(long pos, SeekOrigin o)
            {
            switch(o)
            {
caseSeekOrigin.Begin:
m_Stream.Position = pos + m_DataPos;
                                break;
caseSeekOrigin.Current:
m_Stream.Seek(pos, SeekOrigin.Current);
                                break;
caseSeekOrigin.End:
m_Stream.Position = m_DataPos + m_Length - pos;
    break;
                    }
returnthis.Position;
                    }
publicoverrideint Read(byte[] buf, int ofs, int count
)
{
int toread = (int)Math.Min(count, m_Length -
Position);
return m_Stream.Read(buf, ofs, toread);
                    }
publicoverridevoid Write(byte[] buf, int ofs, int count)
{
thrownewInvalidOperationException();
            }
        }
    }
```

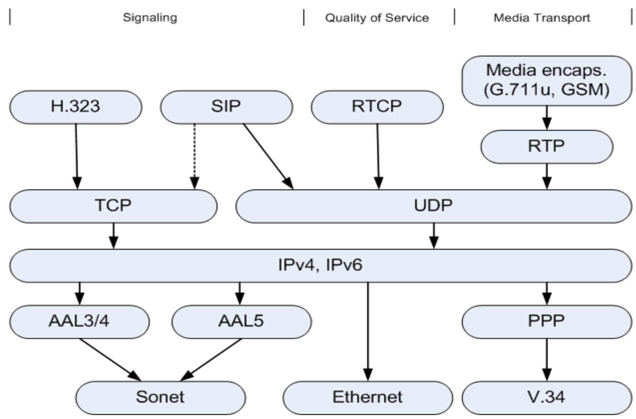

Fig 1: VoIP Architecture

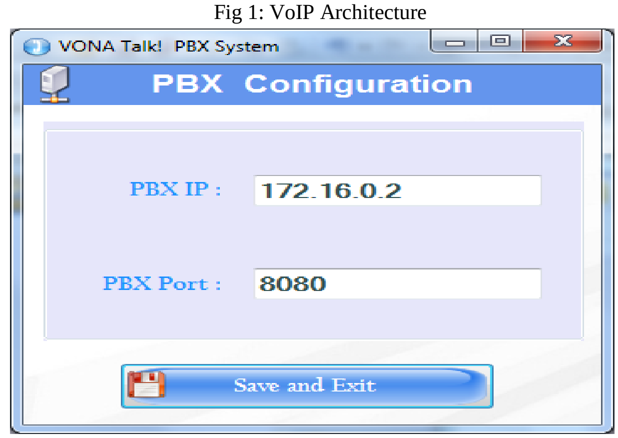

Fig 2: PBX Server

Fig 3: Soft phone Client

5. Conclusion & Future Work:

With the faster pace of development and growing demands of the users for higher quality of voice data transfer at lower bandwidths and better performance over faster growing networks. Integrated services and communications on-the-fly are the demands of industrial, commercial, telecommunications, information technology and the educational sector. With data and voice services being provided with committed quality of service which is the main goal of this research paper. The proposed system has been tested in environment of the campus at Fatima Jinnah Women University and works effectively, fulfilling the needs of the current user base (lecturers/staff/students). However, with networks becoming larger in size and services being offered, the compression techniques being improved at faster pace the existing system can be further improved / refined to meet the user needs.

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