Voice over Internet Protocol with Novel Applications

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Abstract:- Internet Telephony, often denoted as Voice-over-Internet-Protocol (VoIP), has gained more and more attention world-wide during the last decades. Voice over Internet Protocol (VoIP) technology has become a communication alternative with the continuous increasing of Internet bandwidth and rapid advancement of peer-to-peer (P2P) applications.In this paper three types of VOIP are explained: (i) PC to PC : this is the easiest way to use VOIP, which enables you to talk and communicate Voice over Internet with all people over the world. (ii) PC to Phone: which need a gateway that connects IP Network to phone Network. Its uses a device called an ATA (Analogue Telephone Adaptor).The ATA allows you to connect a standard phone to your computer or your Internet connection for use with VOIP. The ATA is an analogue to digital converter. (iii) Phone to Phone: Where you need more gateways that connect IP network to phone networks, more phone networks that connect Telephone set to gateway, and IP network that connect gateway to gateway. This paper also explains Internet Protocol (IP) that VOIP uses to transmit voice as packets over an IP network as follows: H.323 Protocols that provides the technical requirements for voice communication over LANs, while assuming that Quality of Service isn't provided by LANs. Session Initiation Protocol (SIP) standard which is the standard for establishing VOIP connections. This paper also explores Voice XML which is a markup language derived from XML for writing telephone-based speech applications.In this paper also VoiceXML developed environments(Gateways) are explained, where a list of all VoiceXML developers is included, and the most popular VoiceXML development environments (Gateways) are explained in details.Finally, in this paper ten VoiceXML applications are developed.

Keywords:- VOIP , XML, H.323 Protocol, etc.

I. INTRODUCTION

VOICE OVER Internet Protocol (VoIP) is one of the fastestgrowing Internet applications today [1].Voice over Internet Protocol (VoIP) is a technology that allows you to make telephone calls using a broadband Internet connection instead of a regular (or analog) phone line. It is essentially a phone network that uses the Internet infrastructure that already connecting computers all over the world, and lets you make free long-distance calls using your computer.

Some services using VoIP may only allow you to call other people using the same service or a special VoIP phone, but other services allow you to use a traditional phone through an adaptor to call anyone who has a telephone number including local, long distance, mobile, and international numbers.

On implementation of this technology provides the common wiring set up for the computers as well as phone lines for the communication [2].In an effort to offer convenient and secure networking services, researchers have proposed various defensive mechanisms over the past few years, such as intrusion detection systems (IDSs) [3], [4], [5], [6] and prevention mechanisms (PMs) [7], [8], [9], [10].

A. Working of VoIP

Voice over Internet Protocol, is a method for taking analog audio signals, like the kind you hear when you talk on the phone, and turning them into digital data that can be transmitted over the Internet. In simple way VoIP converts the voice signal from your telephone into a digital signal that travels over the Internet.

VoIP can turn a standard Internet connection into a way to place free phone calls, using some of the free VoIP software that is available, bypassing the phone company and its charges entirely.

B. Types of VoIP Calls

1. PC to PC: You need the following tools:
a. PC with sound card.
b. IP Network.
c. IP Telephony software.

    d.   Internet Phone.
    e.   Video optional.

2. PC to Phone and Phone to Phone
For PC to Phone You Need the following tools:
a.   PC with sound card.
b.   Need a gateway that connects IP Network to phone Network.
c.   IP Network.
d.   Phone Network.

For Phone to Phone You Need the following tools:
a.   More gateways that connect IP network to phone networks.
b.   More phone networks that connect Telephone set to gateway.
c.   IP Network that connect gateway to gateway.

Fig.1 Types of VoIP Calls

C. Usage of VoIP
1. Usage of PC to PC
a.   MSN Hotmail Messenger.
b.   Yahoo Messenger.
c.   Windows Messenger.
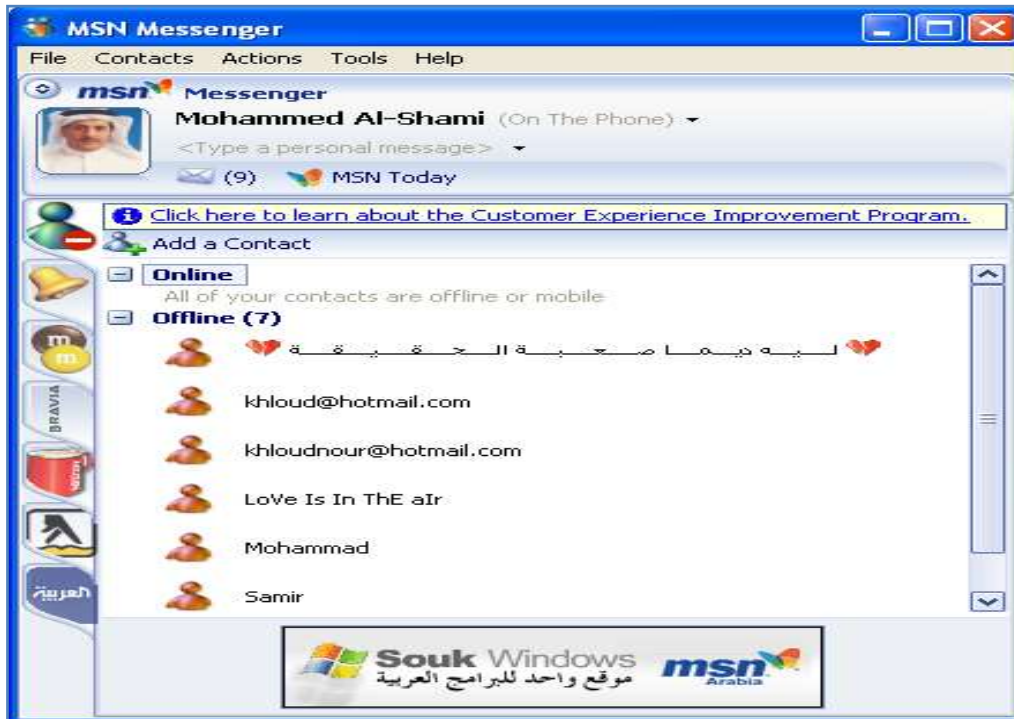**d.**   NetMeeting Messenger.
e.   Pal Talk Messenger.

Fig.2 Usage of PC to PC

2. **Usage of PC to Phone and Phone to Phone**
a. BeVocal Cafe.
b. VoiceGenie.
c. AppBuilder.
d. Audium.
e. TellMe Studio.
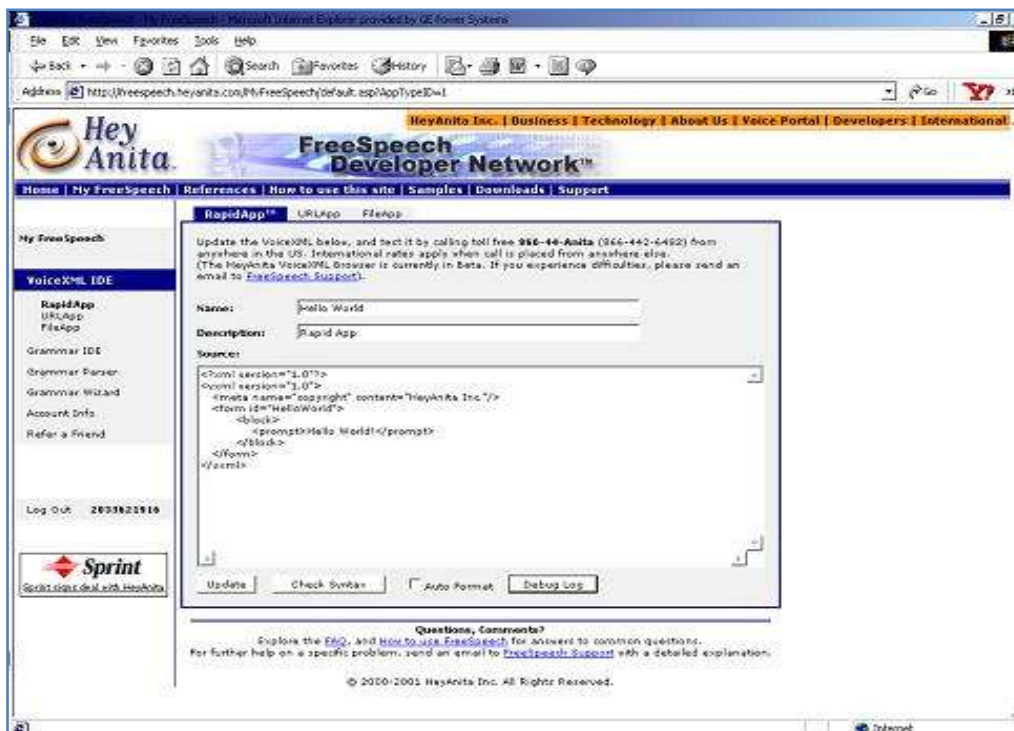f. Vocomo Voice.
g. SyntellectVistaGen XML Edition.

Fig.3 Usage of PC to Phone and Phone to Phone

3. Usage of PC to Phone and Phone to Phone

Fig.4 Usage of PC to Phone and Phone to Phone

D. VoIP Features
    a. **Flexibility:** With VoIP, you can make a call from anywhere you have broadband connectivity.
    b. **Price:** Most VoIP companies are offering minute-rate plans structured like cell phone bills for as little as $30 per month. On the higher end, some offer unlimited plans for $79.
    c. **Equipment:** A broadband Internet connection is required to make VoIP calls. Some services allow you to use a regular telephone, as long as you connect it to an adaptor. Some companies allow you to make calls from a computer or a VoIP phone that doesn't require an additional adapter.
    d. **Whom Can VoIP Call:** Depending upon your service, the call can be made to a local number, a mobile phone, a long distance, or an international number.
    e. **VoIP Provider Call Permit:** Some VoIP providers offer their service for free, some providers charge for a long distance call, Other permit you to call anywhere.
    f. **VoIP Phone Adapter:** You may be able to use your VoIP service wherever you travel as long as you have a broadband Internet connection available.

E. Advantages and Disadvantages of VoIP
Advantages:
    a. Because VoIP is digital, it may offer features and services that are not available with a traditional phone.
    b. If you have a broadband Internet connection, you need not maintain and pay the additional cost for a line just to make telephone calls.
    c. VoIP technology uses the Internet's packet-switching capabilities to provide phone service.
    d. VoIP has several advantages over circuit switching. Packet switching allows several telephone calls to occupy the amount of space occupied by only one in a circuit-switched network.
    e.
Disadvantages:
    a. Some VoIP services don't work during power outages and the service provider may not offer backup power.
    b. It may be difficult for some VoIP services to seamlessly connect with the 911 emergency dispatch center or to identify the location of VoIP 911 callers.
    c. VoIP providers may or may not offer directory assistance/white page listings.

F. VoIP Phone System and Packet Switching

Packet switching opens a brief connection, just long enough to send a small chunk of data, called a packet, from one system to another.The sending computer chops data into small packets, with an address on each one telling the network devices where to send them.Inside of each packet is a payload, which is a piece of the e-mail, a music file or whatever type of file is being transmitted inside the packet.The sending computer sends the packet to a nearby router and forgets about it. The nearby router send the packet to another router that is closer to the recipient computer. That router sends the packet along to another, even closer router, and so on. When the receiving computer finally gets the packets, it uses instructions contained within the packets to reassemble the data into its original state.

Packet switching is very efficient. It lets the network route the packets along the least congested and cheapest lines. It also frees up the two computers communicating with each other so that they can accept information from other computers, as well.Typical Telephone call, using VoIP over a Packet Switched NetworkYou pick up the receiver, which sends a signal to the ATA.The ATA receives the signal and sends a dial tone.You dial the phone number of the party you wish to talk to. The tones are converted by the ATA into digital data and temporarily stored.The phone number data is sent in the form of a request to your VoIP company's call processor. The call processor checks it to ensure that it is in a valid format.The call processor determines to whom to map the phone number, the soft switch connects the two devices on either end of the call. On the other end, a signal is sent to your friend's ATA, telling it to ask the connected phone to ring.Once your friend picks up the phone, a session is established between your computer and your friend's computer. Each system must use the same protocol to communicate.You talk for a period of time. During the conversation, your system and your friend's system transmit packets back and forth when there is data to be sent.The ATAs at each end translate these packets as they are received and convert them to the analog audio signal that you hear. Your ATA also keeps the circuit open between itself and your analog phone while it forwards packets to and from the IP host at the other end.You finish talking and hang up the receiver.When you hang up, the circuit is closed between your phone and the ATA.The ATA sends a signal to the soft switch connecting the call, terminating the session.

G. VoIP Codec

A codec, which stands for coder-decoder, converts an audio signal into a compressed digital form for transmission and then back into an uncompressed audio signal for replay. This is the essence of VoIP.

Digital-to-analog conversion is seen in everything from CD players to cell phones to video game consoles. Codes accomplish the conversion by sampling the audio signal several thousand times per second. It converts each tiny sample into digitized data and compresses it for transmission. When the samples are reassembled, the pieces of audio missing between each sample are so small that to the human ear, it sounds like one continuous second of audio signal.

II.   VoIP Protocols and Standards

A.   H.323 STANDARD

It is the ITU (International Telecommunications Union) standard that vendors should comply while providing Voice over IP service.The first version of H.323 was released in 1996 while the second version came into effect in January 1998.It provides the technical requirements for voice communication over LANs while assuming that no Quality of Service is being provided by LANs.It was originally developed for multimedia conferencing on LANs, but was later extended to cover Voice over IP.It defines four logical components viz., Terminals, Gateways, Gatekeepers and Multipoint Control Units (MCUs). Terminals, gateways and MCUs are known as endpoints.
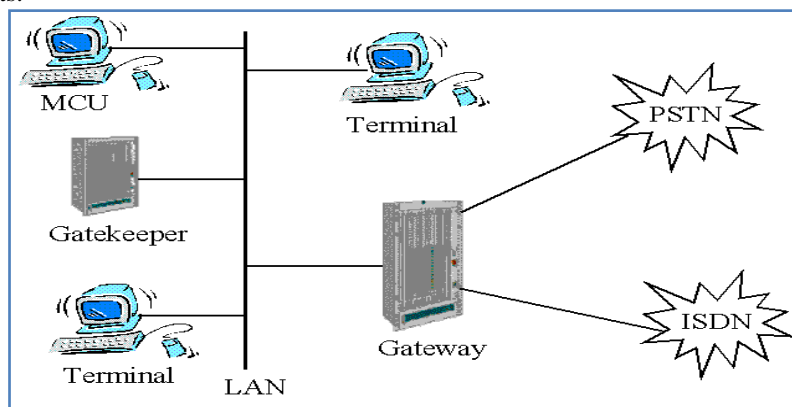

Fig.5 H.323 logical components

B.   H.323 Terminals

These are the LAN client endpoints that provide real time, two way communications.All H.323 terminals have to support H.245, Q.931, Registration Admission Status (RAS) and Real Time Transport Protocol (RTP).H.245 is used for allowing the usage of the channels.Q.931 is required for call signaling and setting up the call.RTP is the real time transport protocol that carries voice packets while RAS is used for interacting with the gatekeeper.H.323 terminals may also include T.120 data conferencing protocols, video codes and support for MCU. A H.323 terminal can communicate with either another H.323 terminal, a H.323 gateway or a MCU.

C. H.323 Gateway

It is the interface between the PSTN and the Internet. It take voice from circuit switched PSTN and place it on the public Internet and vice versa.It is an endpoint on the network which provides for real-time, two-way communications between H.323 terminals on the IP network and other ITU terminals on a switched based network, or to another H.323 gateway.It perform the function of a "translator", and the translation between different transmission formats, such as from H.225 to H.221. It also capable of translating between audio and video codes.Gateways are optional in that terminals, where a single LAN can communicate with each other directly. When the terminals on a network need to communicate with an endpoint in some other network, then they communicate via gateways using the H.245 and Q.931 protocols.

D.   Gatekeepers

It is the most vital component of the H.323 system and dispatches the duties of a "manager". It acts as the central point for all calls within its zone and provides services to the registered endpoints ((A zone is the aggregation of the gatekeeper and the endpoints registered with it).
Some of the functionalities that gatekeepers provide are:
❖   Address Translation.
❖   Admissions Control.
❖   Call signaling.
❖   Call Authorization.
❖   Bandwidth Management.
❖   Call Management.

E.   Multipoint Control Units (MCU)

It is an endpoint on the network that provides the capability for three or more terminals and gateways to participate in a multipoint conference.It consists of a mandatory Multipoint Controller (MC) and optional Multipoint Processors (MP).It determines the common capabilities of the terminals by using H.245 but it does not perform the multiplexing of audio, video and data. The multiplexing of media streams is handled by the MP under the control of the MC.

F.   Call Setup in H.323

The procedure to set up a call involves: Discovering a gatekeeper which would take the management of that endpoint.Registration of the endpoint with its gatekeeper.Endpoint enters the call setup phase.
The capability exchange takes place between the endpoint and the gatekeeper.The call is established.
When the endpoint is done, it can terminate the call. The termination can also be initiated by the gatekeeper.

G. Session Initiation Protocol (SIP)

It is the standard for establishing VoIP connections. It is an application layer control protocol for creating, modifying and terminating sessions with one or more participants.The architecture of SIP is similar to that of HTTP (client-server protocol). Requests are generated by the client and sent to the server. The server processes the requests and then sends a response to the client. A request and the responses for that request make a transaction.SIP has INVITE and ACK messages which define the process of opening a reliable channel over which call control messages may be passed. A SIP makes minimal assumption about the underlying transport protocol.It provide reliability and does not depend on TCP for reliability. It depends on the Session Description Protocol (SDP) for carrying out the negotiation for codec identification. It supports session descriptions that allow participants to agree on a set of compatible media types. It also supports user mobility by proxy and redirecting requests to the users current location.

H. The Services that SIP provide
   a.   User Location: determination of the end system to be used for communication
   b.   Call Setup: ringing and establishing call parameters at both called and calling party.

c.   User Availability: determination of the willingness of the called party to engage in communications.
d.   User Capabilities: determination of the media and media parameters to be used.
e.   Call handling: the transfer and termination of calls.

I. Components of SIP
a.   **User Agents:** A user agent is an end system acting on behalf of a user. There are two parts to it:
User Agent Client (UAC) which is used to initiate a SIP request.
User Agent Server (UAS) which is used to receive requests and return responses on behalf of the user.
b.   **Network Servers:** There are 3 types of servers within a network:
A registration server receives updates concerning the current locations of users.
A proxy server on receiving requests, forwards them to the next-hop server, which has more information about the location of the called party.
A redirect server on receiving requests, determines the next-hop server and returns the address of the next-hop server to the client instead of forwarding the request.

J.SIP Messages
a.   INVITE: for inviting a user to a call.
b.   BYE: for terminating a connection between the two end points.
c.   ACK: for reliable exchange of invitation messages.
d.   OPTIONS: for getting information about the capabilities of a call.
e.   REGISTER: gives information about the location of a user to the SIP registration server.
f.   CANCEL: for terminating the search for a user.

K.SIP Operation
a.   SIP Addressing.
b.   Locating a SIP server.
c.   SIP Transaction.
d.   SIP Invitation.
e.   Locating a User .
f.   Changing an Existing Session

L.Sample SIP Operation
        A SIP client creates an INVITE message for mhalshami@hotmail.com, which is normally sent to a proxy server.The proxy server tries to obtain the IP address of the SIP server that handles requests for the requested domain. Also the proxy server consults a Location Server to determine this next hop server.The Location server is a non SIP server that stores information about the next hop servers for different users.On getting the IP address of the next hop server, the proxy server forwards the INVITE to the next hop server. After the User Agent Server (UAS) has been reached, it sends a response back to the proxy server.The proxy server in-turn sends back a response to the client. The client then confirms that it has received the response by sending an ACK.
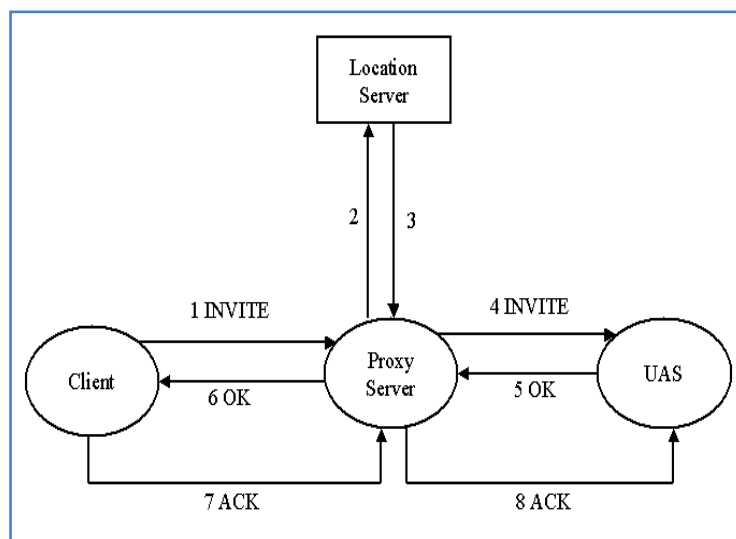

Fig.6 Example of a SIP operation

III. VOICEXML DEVELOPMENTENVIRONMENTS

A. Development Environments

It provide a convenient way to make VoiceXML documents accessible by telephone.

It have many very useful online tools to expedite authoring and debugging of applications.

If we prefer not to develop applications online, then standalone, end-to-end voice application development toolkit are available that simulate the telephone interface.

The best one supports Text-to-Speech (TTS) and automated speech recognition (ASR) to allow you to test application using your desktop's sound output and microphone.

Offer a suite of tools to make VoiceXML development and debugging easier.

B. Development Environments Facilities

a.   Syntax Checkers.
b.   Real time debuggers.
c.   Logs of past incoming phone calls.
d.   Simulators.
e.   Extensive Documentation.
f.   Libraries of audio clips, grammars, and code samples.

C. Voice Gateway

It is highly scalable server software, designed to enable telephone access to enterprise applications.

It integrates speech recognition, text to speech and telephony technologies with a powerful VoiceXML Interpreter to enable voice access to information and services from any telephone.

The Voice Gateway scales to accommodate voice applications of any size, and is capable of supporting millions of users.Some of the known voice gateways are Be vocal, Voice Genie, Telstra Clear, Tellme, Hey Anita and Voxeo.These provide a free development environment. You can run your VXML, ASP, JSP web application files from your own web server and leave the tricky telephony stuff to these voice gateways, whenever one signs up with these voice gateways. If you plan to run a remote based application then they provide the account number and set you point to the URL of your Voice XML application. If you plan to build a localized application then the voice gateways can allow you a certain amount of space on their own servers.

D. Be Vocal Café Developer Workshop

It develop telephone-based speech applications that provide access to web content, where users run these applications by calling them on the telephone.They listen to options and provide input using the spoken word, as opposed to viewing a screen display and entering information with a keyboard or mouse. You use VoiceXML to program your speech applications. VoiceXML interpreter renders VoiceXML documents audibly. VoiceXML interpreter is a telephone-based voice browser.

In contrast, a standard web browser runs locally on your machine, whereas the VoiceXML interpreter is located remotely at its hosting site. And, you use any telephone to access the VoiceXML interpreter that runs your speech application. Be Vocal Café also provides you with a library of reusable VoiceXML components. These include libraries of reusable audio clips and grammars.

E. Be Vocal Café Developer Tools

The **File Managemen**t page lets you upload, view, and delete files that you have stored on the Café server.The **VoiceXML Checker** can check the syntax of your documents. Although it does not guarantee that they will run correctly, it does check for correct syntax, properly nested elements, missing quote characters, and so on.

The **Vocal Player** lets you replay calls that have been made to your application.

The **Log Browser** shows you detailed log files of calls to your application, showing what files were fetched, and values were computed, and so on.

The **Trace Tool** lets you monitor a call while a user is on the phone.
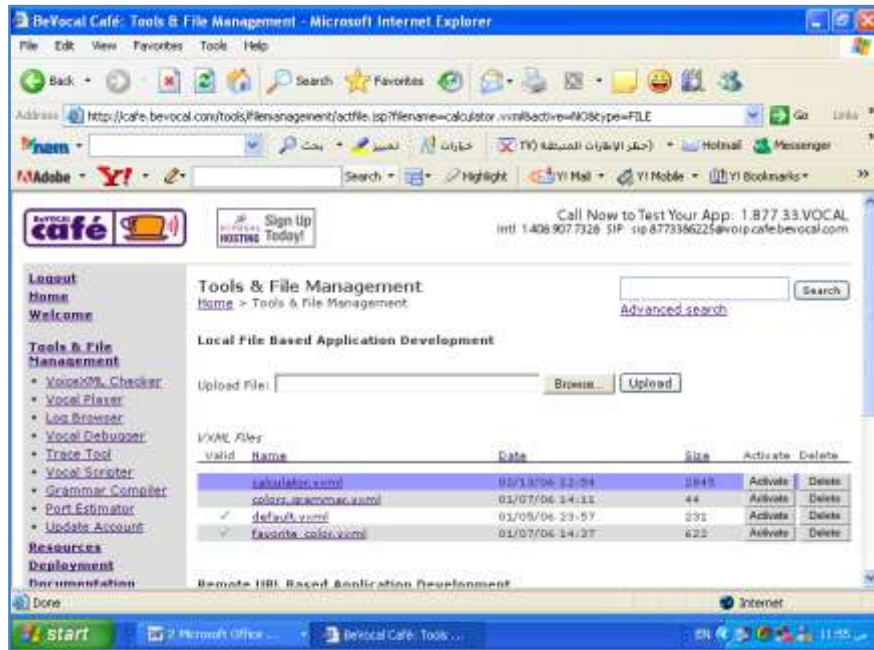
Fig.7 Be Vocal Café Developer Tools

F. Voice Genie Developer Workshop
It is a group of tools, documentation, and other resources that allows you to design, develop and test your own voice-enabled, interactive phone application for free.
Developers are using Voice Genie, with deployed applications.
All you need to develop VoiceXML applications is your web browser, an ordinary phone, some web space and a free Developer Workshop account.
        As a member of the Developer Workshop you can manage multiple extensions and host multiple applications, access call log and trace information on all calls, validate your VoiceXML page syntax on-line, use our grammar tools to learn about and test, link your VoiceXML application to our VoiceGenie VoiceXML Gateway, access discussion forums.

G. Voice Genie Developer Tools
Once you've signed up on Voice Genie Developer Workshop, you're ready to begin building your VoiceXML application.
You can add up to 20 different extensions to call your applications.
        Use Voice XML Validator to check your code and to get a visual representation of your voice application or use the Call Log Explorer to keep track of the calls made to your extensions. Write your VoiceXML application. Upload your VoiceXML application to a web site. Log in to VoiceGenie Developer Workshop.
Create an extension that points to your application:
    a. Click on the "Tools" tab.
    b. Go to the "Extension Manager".
    c. Type the URL for your application in the table.
    d. Click on "Add", and make a note of the extension you're given.
Test your application:
    a. Call either Telephone Numbers, Developer +1 416-736-9731, or Speech Genie +1 416-736-6779 which are listed in the Extension Manager.
    b. When you are asked for your extension, either say it or type it in using the touchtone keypad on your telephone.

Fig.8 Voice Genie Developer Workshop

IV. VOICE XMLVOICE EXTENSIBLE MARKUP LANGUAGEVERSION 2.0

A. History

The origins of VoiceXML began in 1995 as Phone Markup Language (PML) by AT&T.

Lucent and Motorola continued working on PML-like languages.

In 1998, Motorola had developed Vox ML, IBM had developed SpeechML, HP's had developed TalkML and Pipe Beach's had developed Voice HTML.

In 2000, the VoiceXML Forum released VoiceXML 1.0 to the public.

VoiceXML 1.0 was submitted to the W3C as the basis for the creation of a new international standard. VoiceXML 2.0 is the result of this work based on input from W3C Member companies, other W3C Working Groups, and the public.

B. Voice XML Overview

Voice XML is a markup language derived from XML for writing telephone-based speech applications. Users call applications by telephone. They listen to spoken instructions and questions instead of viewing a screen display; they provide input using the spoken word and the touchtone keypad instead of entering information with a keyboard or mouse.

Just as a web browser renders HTML documents visually, a VoiceXML interpreter renders VoiceXML documents audibly. VoiceXML interpreter is a telephone-based voice browser.

VoiceXML documents have web URIs and can be located on any web server. Yet a standard web browser runs locally on your machine, whereas the VoiceXML interpreter is run remotely at the VoiceXML hosting site. And you use your telephone to access the VoiceXML interpreter.

In order to support a telephone interface, the Voice XML interpreter runs within an execution environment that includes a telephony component, a text-to-speech (TTS) speech-synthesis component, and a speech-recognition component.

C. Voice XML interpreter

Voice XML interpreter transparently interacts with these infrastructure components as needed. For example:

Text strings in output elements are rendered using TTS.

a. Connection issues (picking up the incoming call, detecting a hang-up, transferring a call) are handled by the telephony component.

b. Listening to spoken input from the user and identifying its meaning is handled by the speech-recognition component.

A user connects with your application by dialing the appropriate phone number. VoiceXML interpreter answers the call and starts executing your VoiceXML document. Under the document's control, the interpreter may perform actions such as:

a. Sending vocal prompts, messages, or other audio material (such as music or sound effects) to the user.
b. Accepting numeric input that the user enters by DTMF (telephone key tone) signals.
c. Accepting voice input and recognizing the words.
d. Accepting voice input and simply recording it, without trying to recognize any words.
e. Sending the user's information to a web site or other Internet server.
f. Receiving information from the Internet and passing it to the user.

D.   Commercial Aspects of VoiceXML
a. Interactive voice response systems that provide various kinds of information to the clients of an enterprise.
b. Call centers.
c. Telephone-based banking applications.
d. Ticket reservation services.
e. Services such as voice-activated dialing and voice mail services, which are in increasing demand.

E.   Architecture of VoiceXML
a. Web server.
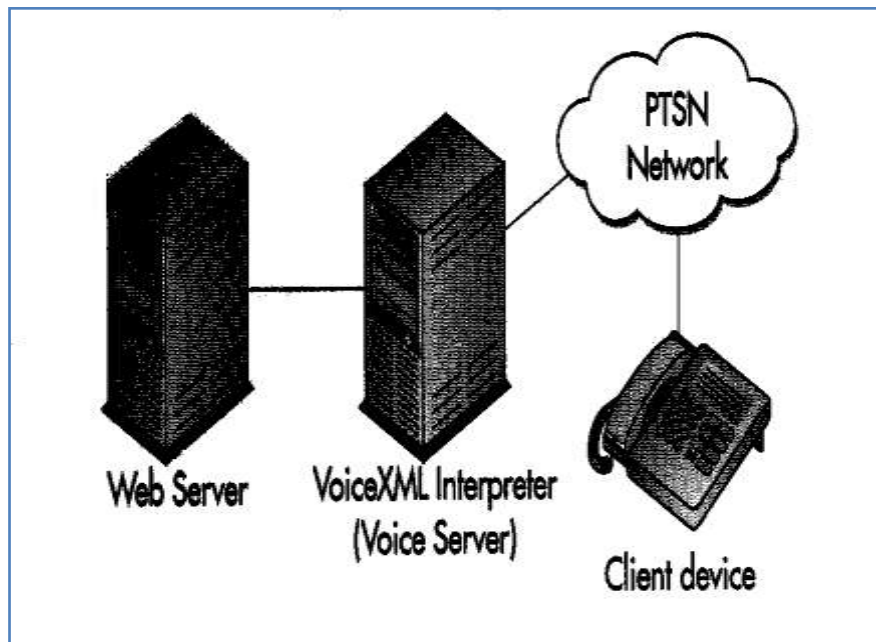b. VoiceXML Interpreter.
c. PSTN network.
d. Client device

Fig.9 Architecture of VoiceXML

VII.   VOICE XML NOVEL APPLICATIONS

A. Electrical Engineering Department Meeting Application.
a. Application Interactions:
Application: Welcome to king Abdul-Aziz University, Electrical Engineering Department; Meetings Schedule. Please choose from the following: (i) For college of engineering Meeting, say college. (ii) For electrical engineering Meeting, say electrical. (iii) For computer engineering meeting, say computer. (iv) For Electronics & Communication engineering meeting, say electronics or communications. (v) For power and electrical machines engineering meeting say power.
User: Computer
Application: The computer engineering meeting will be held every week on Sunday at 9:30 AM , thank you for using Electrical Engineering meeting program.

B. Post Graduate Admission Application.

a**. Application Interactions:**

Application: Welcome to king Abdul-Aziz University; the conditions of postgraduate admission program. Please choose from the following: (i) for diploma degree admission ,say diploma. (ii) for master degree admission, say master. (iii) For PHD degree admission, say PhD.

User: Master

Application: The conditions of Admission for the degree of master will be as follows: (i) Applicant must have atleast a grade of very good in the bachelor degree. (ii) Applicant must be Saudi citizen or has official scholarship from KSA if not Saudi citizen. (iii) Applicant should have bachelor degree from Saudi university or a university internationally recognized. (iv) Applicant must have well behavior also physically and clinically fit. (V) Applicant must attach two recommendations from two of his teachers. (vi)Applicant must attach approval letter from his work if the applicant is an employee.Thank you for using postgraduate Admission Program.

C.Courses Application

a. Application Interactions:

Application: Welcome to King Abdul-Aziz University, Electrical Engineering Department ,Master degree Courses Program that will be provided for first semester of the education year 1427 , 1428 H. Please choose from the following: (i) for computer engineering courses, say computer. (ii)For electronics & Communications engineering courses, say electronics or communications. (iii) For power & electrical machines engineering courses, say power. For obligatory Mathematical courses, say mathematics.

User: computer

Application: The computer engineering courses will be 540 that is titled programming language, and 547 that is titles computer networking. Thanks you for using courses program.

D. Courses Scores Application

a. Application Interactions:

Application: Welcome to King Abdul-Aziz University, electrical engineering department, Master degree courses program, for the first semester of the educational year 1427-1428 H. Please say your student number to get your scores.

User: 0393034

Application: Student number 0393034 .Your Name is Mohammed Hassan Al-Shami and your scores are as follows: In 540 programming language course you got B exactly 82 , and in 547 computer networking course you got also B exactly 84. Thank you for using Mohammed courses scores program.

E.Weather ForecastApplication

a. Application Interactions:

Application: Welcome to Weather Forecast Program. Choose Jeddah, Riyadh, taif, medina, Dhahran, Albaha or najran.

User: Jeddah

Application: Insha'Allah the weather of Jeddah mostly sunny today with highs in 36 degree and lows tonight in the 25 degree, on the cost winds will be southeasterly 10 to 15 knots, red sea will run 2 to 4 feet,winds and seas will be 3 feet high near north cost, high tide at 1 p.m., low tide at 12:30 a.m.

F. CalculatorApplication

a. Application Interactions:

Application: Welcome to Mohammed calculator. Choose add, subtract, multiply or divide.

User: add

Application: Let's add two numbers. What's the first number?

User: 25

Application: And second number?

User: 55

Application: 25 plus 55 equals 80. Thank you for using calculator program.

G. **Factorial Number Application**

Application: Welcome to Factorial Program to compute the factorial number. Please say the number that you want its factorial.

User: 4

Application: Factorial of 4 is 24. Thank you for using Factorial Program.

H. Square Number.
a. Application Interactions:
Application: Welcome to Square Program to compute the Square for any number. Please say the number that you want it's square.
User: five
Application: The Square of 5 is 25. Thank you for using square program.

I. Triangle AreaApplication
a. Application Interactions:
Application: Welcome to triangle area program to compute the area for any triangle whose base and height are known. Please say the base Length of the Triangle?
User: Eight
Application: And the height Length of the Triangle?
User: Six
Application: The area of the triangle equals 24. Thank you for using Triangle Program.

J. Circle AreaApplication
a. Application Interactions:
Application: Welcome to circle Area program to compute the Area for any circle. Please say the radius length of the circle that you want its area?
User: Seven
Application: The Circle Area equals 153.9972. Thank you for using Circle Area Program.

VI.   CONCLUSION

In this paper, we provided an overview of the voice over Internet protocol and we showed the importance of VOIP in the communications technology. We also provided an overview of the voiceXML development environments (Gateways), and we showed its importance in developing and debugging VoiceXML applications.We also provided an overview of the VoiceXML with its items and we showed its importance in the voice applications. We designed and implemented electrical engineering meeting application that provides the users with meeting schedule and its time, Post graduate admission application that provide the users with admission procedures for diploma, master or Phd, courses Application that provides the users with the courses that could be taught in the assigned semester, Courses Scores Application that provides the users with the scores obtained in the courses as result of the assigned semester, Weather Forecast Application that provide the users with the Weather Forecast for Jeddah, Riyadh, Medina, Taif ,Dhahran, albaha or Najran city, calculator application that add, subtract, divide or multiply any two numbers and provides the user with answer, square number application that provides the user with square for any number, Mohammed triangle application that provides the user with area for any triangle, circle application that provides the user with area for any circle.

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