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Customize Approach for Detection and Prevention of Unsolicited Call in VoIP Scenario: A Review

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ABSTRACT: Voice over internet protocol is an artistic technology in Internet today. And it's very powerful over the PSTN . As VoIP is more flexible and low cost compare to PSTN, many marketing agencies uses it for marketing call or for message broadcasting calls and messages which make customary users disgruntled. This problem is known as unsolicited bulk calls, called spam over VoIP . In Cellular technology, One method has been implemented to protect the Spam call, called Do Not Disturb(DND).Which is complex to accomplish in VoIP Scenario. This papet presents the fundamentals of SIP(Session Initiation Protocol). SIP is the one of the protocols for the VoIP application which has been text based open protocol. Proposed works on a survey on detection and prevention on spam over VoIP. Main focus of proposed work is to provide a mechanism to protect spam call in VoIP scenario. Proposed work contains an innovative solution, which would combined the multiple spam improvement techniques in order to get maximum success rate for blocking spam call**.**

KEY WORDS:- UNSOLICITED CALL, SIP,VOIP.

I. INTRODUCTION

In Modern Society , the internet users have increased. So , in present the telephony has developed to an universal service. Persons are available at anytime and any where . So this service uses the VOIP technology.

VOIP stands for Voice Over Internet Protocol. The VOIP is also called IP telephony ,is the real-time transmission of voice signals using internet protocol over public Internet. Means that , VOIP converts the voice signals from over telephone into digital signal that travels over the internet . VOIP is reduces the cost of communication and communication hardware typically with the PSTN. Most People are aware of the VOIP through the Skype consumer telephone service.[9][10]

SIP(Session Initiation Protocol) is an application layer protocol which provides session management between VoIP client and server. SIP does not carry any voice or video data itself, it allows 2 end points to setup connection transfer their traffic between each other via RTP(Real-Time Transport Protocol). SIP provides mechanism for establishing calls over IP. SIP is Used to setup , modify and terminate the VOIP. SIP is more suited for the Internet. It is uses the two party and multi party sessions. Other application is file transfer, online games, presence information, instance massages, video conferences.[11]

This Protocol used in VOIP.VOIP is technology that allows the telephone calls using internet. In VOIP, conference can be set up with multiple users communicating in real-time. So VOIP compresses the data packets during the transmission , more calls can be handles on one access time. In VOIP, user can also transfer the images ,text along voice and video. Security is also a very important issues in VOIP. VOIP is low cost compare to the PSTN so the aggressor is use for the massages and advertisement ,it make the normal users unhappy. This problem is known as the unsolicited calls in VOIP scenario.

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The goal of this report to analyze and survey the various techniques to detect and prevent unsolicited calls in VOIP. To discuss the qualities and issues of each techniques. Also discuss on the SIP and VOIP call establishment.
The rest of the paper is organized as follows. Section II gives details about Session Initiation Protocol. Section III Describe study about literature survey on various available schemes followed by detailed comparative study in section IV. Section V gives Conclusion. Last section contains list of references used.

II. SESSION INITIATION PROTOCOL

SIP is an application layer protocol that manages, establishes and terminates the call. It can create two party, multiparty or multicast sessions. SIP is text based protocol and transfer different types of payload with different encoding. SIP supports variety of services, including the locating the callee and determining the callee's capabilities.

SIP entities are registrar, location server, proxy server and endpoints(soft phones or physical devices).The proxy server, redirect server, registrar may be combined otherwise may be independent entities. Endpoints communicate with registrar to indicate their presence. This information stored in location server, A user may be registrar via multiple endpoint simultaneously .[12][13]

There are minimum three separate activities share by the SIP based communication. They are follows:-
SIP provides the basic signaling between end points to setup the session. SIP uses the session Description protocol(SDP) to classify the nature of the communication utilize within session. SIP uses the suitable protocol to convey information in the session. The SIP architecture comprises five component which is SIP User Agent(UA),Proxy server, redirect server, location server, registrar server.[14]

A.SIP User agent (UA):-

User Agent may be hardware device or software. The mainly call the back-to-back User Agent operators between both end points of a phone call or a communication channel into two call. B2BUA provides functions are call management, network , interworking, hiding of network.

SIP user agent is the combination of UAS(User Agent Server) and UAC(User Agent Client) .

User Agent Server:-(UAS)

When responding to request for service is known as UAS. UAS responsible for handling request from another end point .UAS may multiple response to the User Agent Client, not necessarily a single response communication between UAS and UAC is client/server.

User Agent Client:-(UAC)

When responding to request for service is known as UAS. UAS responsible for handling request from another end point .UAS may multiple response to the User Agent Client, not necessarily a single response communication between UAS and UAC is client/server.

B. Proxy server:-

Proxy server can provide three services
-Access to a SIP network
-call routing including URI translator
-registration

Proxy server is calls are routed within a SIP VOIP network. Proxy server are intermediate ending that acts as both server and client for the purpose of making request on behalf of other clients. It is also useful for enforcing policy. Proxy server are elements that route SIP request to UAS and SIP response to UAC.

In figure we show that the operation of proxy server in SIP. When the proxy server receive request to the UA, inviting to another party to join this session. Proxy server

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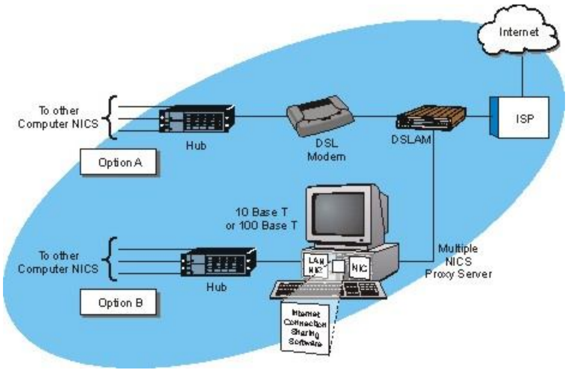

Figure 2.1: Proxy server process[14]

forward this invitation to the UA. If UA is unavailable , then proxy server will direct connect to the second UA or it connect may be forward via one or more proxy servers.

C. Redirect sever:-

Redirect server is a user agent server that generates 3XX responses to the request it receives ,directing the client to URIs .It does not initiate any action . It only provide information.

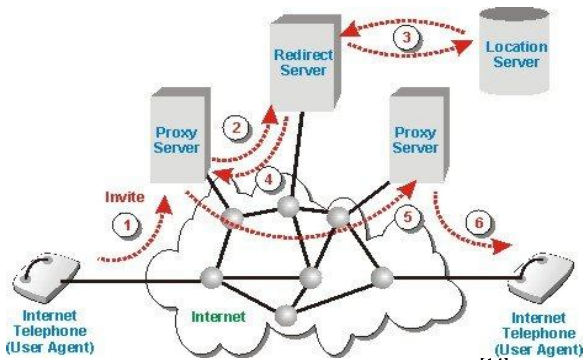

Figure 2.2 : Redirect server process[14]

In figure we show that the operation of redirect server in SIP. Proxy server receives the request of client that specifies that the SIP address of the recipient's address that locate in redirect server. Then the redirect server look at the SIP address in the request and fetch from the database of the current location information of address. then the redirect server will response to the proxy server and proxy server is redirect the particular destination.

D. Location server:-

Location server provides information regarding the location of resources that are locate with in network. It is a database that maintain data of register user.

In figure we show that the user agent request for the communication session to the another user. First this request is received to the proxy server, proxy server send the location query to

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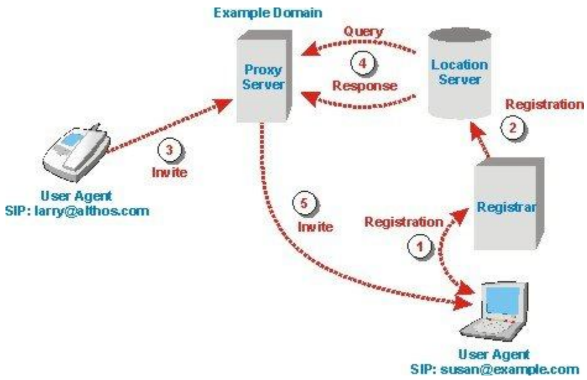

Figure 2.2 : Location server process[14]

the location server of the address of SIP. Location server returns the last location address of SIP. Then proxy server forwards the invite request to the other user.

E. Registrar server:-

Registrar is a server that accepts the register request and place the information of registration, in the database called location service.

In figure 2.2 below we show that the operation of registrar server that is used to gather and stored registration data into a database to provide location service. UA is connected to the internet. Internet telephone had been already programmed to register with a specific register services on the detection of new services. When UA is send to the registration information to registrar, the registrar updates the location services database .Many ways in which the UA may discover a registrar

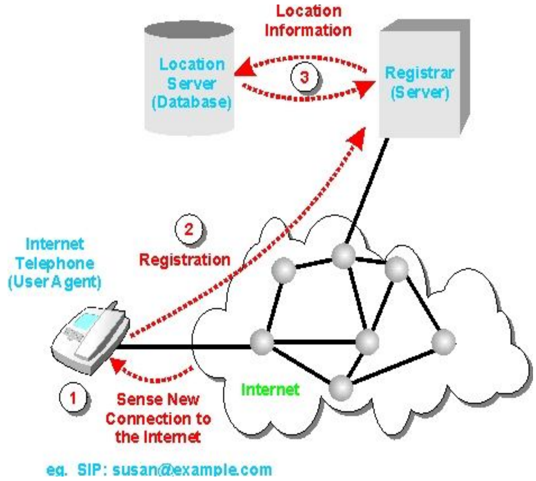

Figure 2.2 : Registrar server process[14]

III. SURVEY ON UNSOLICITED CALLS IN VOIP SENARIO USING VARIOUS METODs

Various researchers give different schemes and views for Spam in VOIP by using Different algorithms.

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Authors of [7] presented a method Anti-SPIT, which contains that the differentiation between the normal calls and SPIT calls. The Proposed structure can use four features that are very effective in VOIP Spam detection process.
In the author the most important features is call duration. In VOIP spam, call duration is very short. And normal call duration is go on.

So the equation is :

Call duration = sum of call duration in period time/number of calls in the same period time

The Second feature is call rate. VOIP spam call usually sends many calls in a certain period of a time while the normal user send few calls in the period of time

The third feature is callee'IDs, IPs and domains. This provides the 8 combinations that can find the spam call. Many calls sends with same ID with different IPs so it could be doubtful. The last feature is patterns .It is checking like call starting time, period, duration etc. Above futures will calculate and it will combined then one could get reputation value .The caller information is compared with backlist data if it is found then it will be blocked otherwise the reputation value will calculated and it will compared to threshold if it reputation value<threshold then it will be in block list. Otherwise it will allow after ending the call the user can send the feedback of the call and if it was spam call then it will be block list.

Observation:-

Reputation of call is calculated as per various criteria like call duration, call rate, IP domain and call pattern. It is not full proof solution. Sometime genuine caller may also consider as spam call. So there are few chances of false positive.
The researchers of[8] proposed scheme is based on six features to detect the spam call. features are call rate , call interval, simultaneous calls, diversity of callers, call duration and error rate. Call rate and call duration features are used to the above anti spit technique. The AIM of author to decrease the False Negative and increase the True Positive.

call interval:-

the inter time distance for the calls generated by the spam VOIP usually use the regular scheme. So their call differences is low.

Simultaneous call:-

If simultaneous calls are occurs , then exactly determine the caller is suspicious.

Diversity of callers:-

SPITTER attempts to cover a large amount of different callees in short time, but generally the normal users can not use this scenario and if it is and it is very rare.

Error rate:-

SPITTERS occurs high volume of SIP errors including CANCLE packet and 404 errors.
In this author first it will check the incoming call within the list. this will increase the detection speed by passing the feature extraction module which takes more time. so, it has to be performed with extra accuracy.
After checking the list we can multiply by the weights that are gained by PCA or LDA approach to features and calculate total value. This value compared with a threshold. If value>threshold then the call is doubtful and then more analysis then we play the CAPTCHA for him in order to answer the question. If answer is wrong the call is dropped and answer is true then send the call.

Observations:-

Some Parameter used in system like SPIT threshold ,must requirements and it can not be determined automatically . Two algorithm and six parameter is used for this method

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IV. COMPARATIVE STUDY

Individual Criteria → Providers ↓	Spam Detection\Reduction	Spam Prevention	Dummy Directory Number	Black \White List	Captcha	Content /based Detection	Call Duration/Behaviour	Social Reputation/reliability	Algorithm proposal Shown	Experimental Setup	Limitation / Enhancement
[1]	D	Y	N	X	Y	X	N	N	N	Y	Can't work with SIP Protocol
[2]	D	Y	N	Y	N	Y	N	N	N	Y	----------
[3]	D	Y	N	N	N	N	Y	Y	N	Y	User may accidentally marks his friend as a spammer.
[4]	R	N	Y	Y	Y	N	N	N	Y	N	Selection of the dummy directory numbers Not deployed in a real VoIP environment.
[5]	D	Y	N	N	Y	Y	N	N	N	Y	Voice CAPTCHAs for those with a visual impairment.
[6]	R	N	N	N	N	N	Y	Y	Y	Y	SPIT threshold not be determined automatically.

Table I Comparative study

The table-I shows above gives detailed comparison about the various schemes proposed by a researcher. The table gives the description about the basic technique used with the benefits that researcher gets as well as the limitations found in schemes.

V. CONCLUSION

In this paper, we studied various research proposals on VoIP Spam, using SIP(Session Initiation protocols) to decrease the unsolicited calls in VoIP scenario . We also made a comparative table which can be used by layman to have handy information about the proposals. Some of the proposals are based on call duration, spam prevention,captcha and spam reduction techniques. We believe that security for Unsolicited call is an area packed of challenges and of vital significance, and many research problems are yet to be identified.

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