International journal of Advances in Computer Networks and Security

A survey on Fax over IP using T.38 over SIP

Disha S.
Department of Information Science and Engineering
M S Ramaiah Institute of Technology
Bangalore, India
sdish13@gmail.com

Sumana M.
Assistant Professor
Department of Information Science and Engineering
M S Ramaiah Institute of Technology
Bangalore, India

***Abstract* - Facsimile (fax) transmission is very important to the world of business. In a world where a significant percentage of long distance Public Switched Telephone Network (PSTN) traffic is composed of fax, great savings in toll charges are possible by utilizing IP networks instead. This paper is a brief survey on Real time Fax over IP and includes details on protocols like session initiation protocol which is responsible for session establishment and T.38 protocol which is mainly responsible for real time fax transmission between IP fax machines. It also deals with Session description protocol responsible for describing the multimedia session which is used in conjunction with session initiation protocol.**

***Keywords* - Fax over IP, T.38, T.30, Session Initiation Protocol, Session description protocol, Proxy Servers, User agent client, Redirect Server.**

I. INTRODUCTION

Fax over Internet Protocol (FoIP), or IP faxing, has been there for many years. The latest generation of FoIP systems combines the benefits of traditional faxing with the cost savings of Internet transmission methods. FoIP is the method of sending faxes over the internet. It changes the transmission medium of faxing in the same way that VoIP (Voice over Internet Protocol) changes the transmission medium of a phone call. For real time IP fax session, session initiation protocol is used for session signaling as it is simpler and more efficient.

The rest of the paper is organized as follows: Section II deals with concepts involved in Fax over IP. Section III discusses about T.38 fax protocol. Section IV is on Session Initiation Protocol. Section V deals with the session description protocol used to describe multimedia sessions. Before conclusion, I have proposed the system that would be developed.

II. FAX OVER IP

Fax over IP (FoIP) refers to the transport of faxes over IP networks, using the protocols (T.37, T.38, SIP, and H.323), transmission methods (Real Time, Store and Forward) and other fundamental components that help in the delivery of faxes over the internet (internet faxing) or within a corporate intranet. There are currently two transmission methods and both are standards specified by the ITU. Recommendation T.37 specifies store and forward, non-real time techniques for sending facsimile with legacy equipment connected via a gateway to a packet network. Recommendation T.38 specifies a real-time operation which does not use voice band data transport, but instead uses packetized information to carry both the handshake sequences and the digitized image data itself between the terminals. This paper mainly focuses on real time IP fax session using T.38 protocol.

In store-and-forward approach which makes use of T37 protocol, fax information is transferred from one fax server (gateway) to another fax server as e-mail attachments. The drawback here is that the fax machines do not exchange information in real-time, so it does not appear like a traditional fax session. The machines can't discuss their capabilities (like paper size, color) and the sender does not receive instant confirmation that each page has been received.

In case of real-time IP faxing which makes use of T38 protocol, fax information is transferred between two machines as IP data packets using a high-level Internet Protocol such as TCP or UDP. These protocols allow for real-time connections that allow the fax machines exchange their capabilities. A real-time IP fax is just like a traditional phone-line fax.

A. Real time Internet Fax session

The alternative to non-real-time fax over IP networks is T.38 protocol. The T.38 protocol gives the look and feel of real time facsimile by emulating the handshake activities of the T.30 protocol on the packet network side. Its basic idea is fax demodulation by a T.38 gateway at the source, packetization of all relevant handshake exchanges, sending of the IP packets across the network and remodulation of the analog line by the receiving T.38 gateway from the information carried in the packet data. The traditional fax session is described below.

Fax machines are digital in nature. But phone lines are analog. So fax machines use a protocol called T.30 (fax protocol) to encode digital information into analog signals on the sending end and decode those analog signals back into digital information on the receiving end. Once each machine knows the other's capabilities, the sending machine scans the page and produces a series of bits (1s or 0s) that represent the black and white areas of the page in digital form. It then converts those bits into analog signals for transmission over the phone line. On the other end, the receiving machine decodes the page data back into digital form, reads the bits and prints out the page based on the instructions provided by those bits.

In case of real time fax session, FoIP uses the same method of compressing and interpreting image data as described above but it uses a different protocol for transmitting the data. The protocol that enables real-time faxing over the internet is the T.38 protocol. T.38 converts traditional fax data into an Internet-friendly format. It's basically a method of packaging T.30 fax signals and data as IP packets on the sending end and turning those IP packets back into T.30 signals and data on the receiving end as shown in Fig.1.

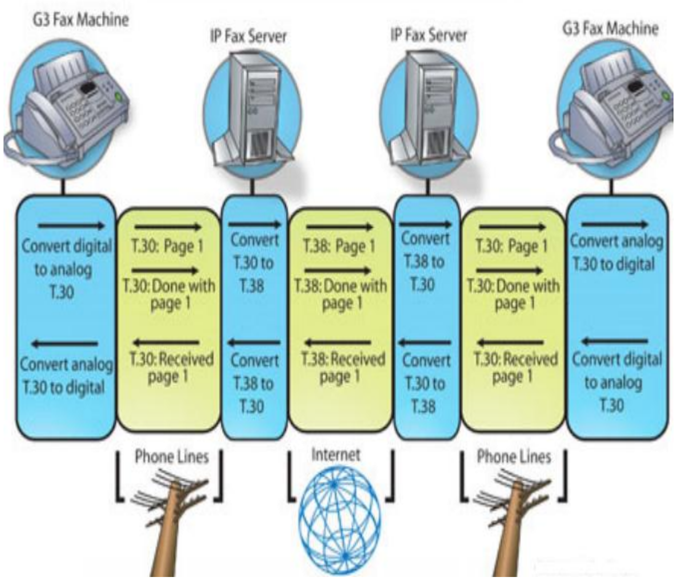

Fig. 1.Real time IP fax session [5]

B. *Communication between Internet Aware Facsimile devices*

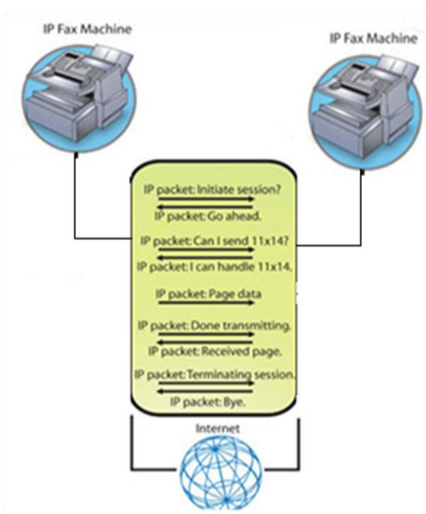

Fig. 2. Communication between Internet Aware Facsimile devices

An Internet Aware Facsimile (IAF) device is a facsimile machine that can access internet directly. Unlike in real time IP fax session which involves telephone lines, here when two IAF's communicate with one another, there is no need for conversion from T.30 to T.38 and vice versa.

There is only the need of T.38 protocol as it is needed for real time IP fax session. This is the concept that will be used where the two devices are Multi Function Printers (MFPs) which are connected to the internet. For session establishment, SIP protocol is used. Finally the fax message received by the other MFP is printed as shown in Fig.2.

III. T.38 PROTOCOL

T.38 is an International Telecommunication Union (ITU) recommendation for allowing transmission of fax over IP networks in real time. It is a protocol that describes how to send a fax over a computer data network. It is needed because fax data cannot be sent over a computer data network in the same way as voice communication. T.38 is not a call setup protocol, thus the T.38 devices need to use standard call setup protocols to negotiate the T.38 call like H.323, SIP and H.248 (Media Gateway control Protocol).

The ITU-T Recommendation T.30 defines procedures for the transmission of Group-3 fax over the Public Switched Telephone Network (PSTN). This includes the end-to-end capabilities negotiations between fax terminals. In order for fax signals to traverse an IP network, gateways are employed to convert between T.30 and T.37 or T.38 protocols for non-real-time and real-time fax, respectively. Recommendation T.37 deals with transmission of fax in a store-and-forward or non-real-time manner. Recommendation T.38 specifies packet format for messages and data exchanged between T.38 gateways on IP networks in real time. It also specifies the exchange of messages and data to other Internet Aware Fax (IAF) devices.

A fax call in conventional Group 3 facsimile is completed in five phases as seen in Fig.3:

1. Phase A - Call establishment
2. Phase B - Attributes, capabilities, and control signaling
3. Phase C - Single-page fax transmission
4. Phase D - End-of-page signaling and multipage notification
5. Phase E - Call termination

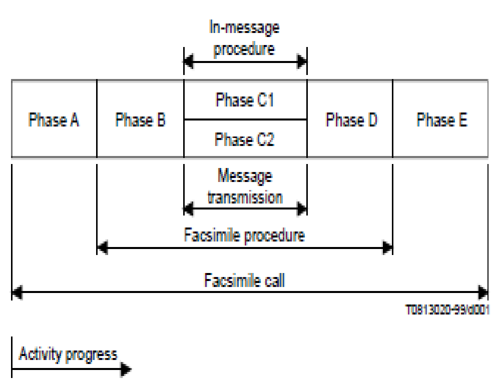

Fig. 3. Time sequence of a facsimile call [3]

Phase A is the Call Establishment phase. In this phase, a CNG is transmitted from the transmitting fax terminal. The T.38 gateway serving the sender demodulates this signal and sends a CNG indicator packet to the gateway serving the receiver, which remodulates a CNG signal on to the receiving fax terminal. In a similar manner, a CED is transmitted back from the receiving terminal followed by a Digital Identification Signal (DIS). This establishes a fax call. Phase B handles the control and capabilities negotiation between the sending and receiving terminals beginning with a Digital Command Signal (DCS) and followed by possibly many optional signals. Phase C is the transmission and corresponding acknowledgment of an actual fax page. Phase D is invoked if there are multiple pages to send. A Multi Page Signal (MPS) indicates a following page, whereas an End of Page (EOP) indicates the end of the last page. Finally, fax transmission is terminated with a Disconnect (DCN) signal in Phase E [3].

Transmissions between T.38 gateways are done with two types of Internet Fax Protocol (IFP) packets. The T.38 packet element has the format shown in Fig.4. The Type field is specified as T30_INDICATOR when the IFP packet carries fax control signals such as Calling Tone (CNG) or Called Station ID (CED). The Type field is specified as T30-DATA when the data field holds fax image data.

Type	Data

Fig. 4. Format of T.38 packet

The IFP allows TCP or UDP to be used as the transport protocol. The protocol is layered, and therefore, whether encapsulation is done using either of these, the message exchange is the same. If TCP is used as shown in Fig.5, the IP payload is only the TCP header and the concatenated IFP packet. When UDP is used as in Fig.6, the payload contains a UDPTL (UDP Transport Layer) header. This UDPTL header contains sequence numbers for packets arriving at the receiving gateway. This is done because UDP packets may traverse different paths and not arrive at the receiving T.38 gateway in the same order in which they were sent. The rest of the payload is the IFP packet to be sent. There are also optional provisions for one or more Forward Error Correction fields. Optional redundant messages can also be included to safeguard against possible loss.

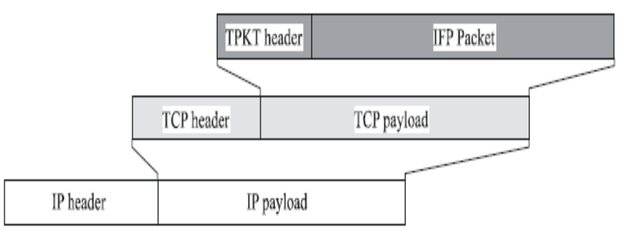

Fig. 5. Layered model of IFP/TCP/IP packet [2]

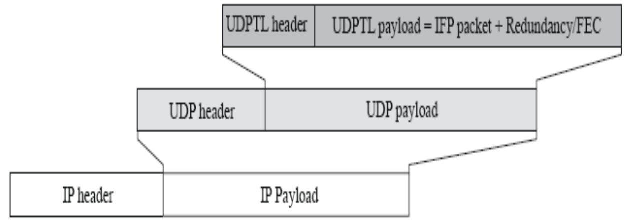

Fig. 6. Layered model of IFP/UDPTL/UDP packet [2]

IV. SESSION INITIATION PROTOCOL

Session Initiation Protocol (SIP) is an application layer control protocol that can establish, modify and terminate multimedia sessions. It incorporates elements of two widely used Internet protocols namely HTTP (Hyper Text Transport Protocol) used for web browsing and SMTP (Simple Mail Transport Protocol) used for e-mail. From HTTP, SIP borrowed a client server design and the use of uniform resource locators (URLs). From SMTP, SIP borrowed a text-encoding scheme and header style. For example, SIP reuses SMTP headers such as To, From, Date and Subject.

SIP supports five facets of establishing and terminating multimedia communications:

1. **User location**: determines the end system to be used for communication.
2. **User availability:** determines the willingness of the called party to engage in communications.
3. **User capabilities:** determines the media and media parameters to be used.
4. **Session setup:** establishment of session parameters at both called and calling party.
5. **Session management:** includes transfer and termination of sessions, modifying session parameters and invoking services.

SIP is a component that can be used with other IETF (Internet Engineering Task Force) protocols to build a complete multimedia architecture. These architectures include protocols such as the Real-time Transport Protocol (RTP) for transporting real-time data, the Real-Time streaming protocol (RTSP) for controlling delivery of streaming media, the Media Gateway Control Protocol for controlling gateways to the Public Switched Telephone Network (PSTN) and the Session Description Protocol (SDP) for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users [1].

A. SIP Entities

1. **User Agent**: A logical entity that can act as both a user agent client and user agent server.

a) User Agent Client (UAC): a user agent client is a logical entity that creates a new request. The role of UAC lasts only for the duration of that transaction.

b) User Agent Server (UAS): a user agent server is a logical entity that generates a response to a SIP request. The response accepts, rejects or redirects the request. This role lasts only for the duration of that transaction.

2. **Proxy Server**: An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity “closer” to the targeted user.

3. **Redirect Server**: A redirect server is a user agent server that generates 3xx (redirection) responses to requests it receives, directing the client to contact an alternate set of URIs.

4. **Registrar**: A registrar is a server that accepts REGISTER requests and places the information it receives in those requests.

B. SIP call flow

The basic functions of SIP include location of an end point, signal of a desire to communicate, negotiation of session parameters to establish the session and teardown of the session once established as depicted in Fig. 7.

In this example, Alice uses a SIP application on her PC (referred to as a softphone) to call Bob on his SIP phone over the Internet. Also shown are two SIP proxy servers that act on behalf of Alice and Bob to facilitate the session establishment. Message is labeled as ‘F’.

Registration is a common operation in SIP. Registration is one way that the biloxi.com server can learn the current location of Bob. Bob’s SIP phone sends REGISTER messages to a server in the biloxi.com domain known as a SIP registrar. The REGISTER messages associate Bob’s SIP URI (sip:bob@biloxi.com) with the machine into which he is currently logged. The registrar writes this association, also called a binding, to a database, called the location service, where it can be used by the proxy in the biloxi.com domain.

The call flow is as follows:

1. Alice calls Bob using his SIP identity, a type of Uniform Resource Identifier (URI) called a SIP URI. It has a similar form to an email address, typically containing a username and a host name. In this case, it is sip:bob@biloxi.com, where biloxi.com is the domain of Bob’s SIP service provider. Alice has a SIP URI of type sip:alice@atlanta.com.
2. Since the soft phone does not know the location of Bob or the SIP server in the biloxi.com domain, the soft phone sends the INVITE to the SIP server that serves Alice’s domain.
3. The proxy server receives the INVITE request and sends a 100 (Trying) response back to Alice’s soft phone.
4. Bob’s SIP phone receives the INVITE and alerts Bob to the incoming call from Alice so that Bob can decide whether to answer the call, that is, Bob’s phone rings. Bob’s SIP phone indicates this in a 180 (Ringing) response, which is routed back through the two proxies in the reverse direction.
5. Bob decides to answer the call. When he picks up the handset, his SIP phone sends a 200 (OK) response to indicate that the call has been answered. The 200 (OK) contains a message body with the SDP media description of the type of session that Bob is willing to establish with Alice.
6. The biloxi.com proxy server would then receive and proxy the ACK, BYE and 200 (OK) to the BYE.

Additional operations in SIP, such as querying for the capabilities of a SIP server or client using OPTIONS or canceling a pending request using CANCEL can be performed.

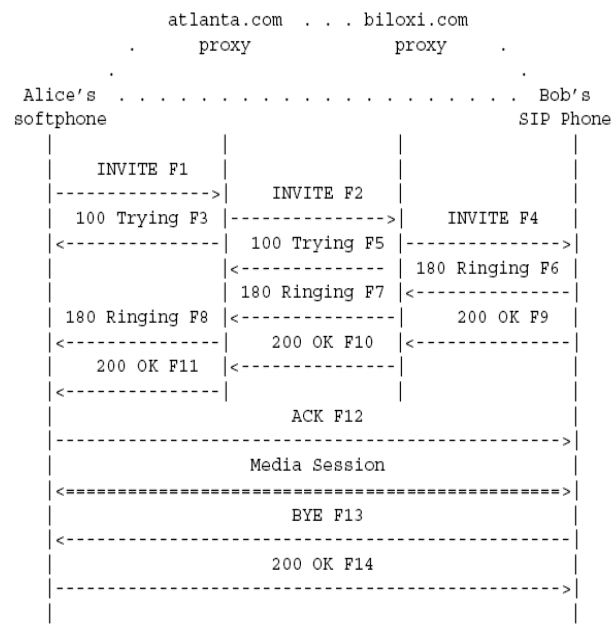

Fig 7: SIP session setup example

C. SIP requests

A SIP transaction between a client and a server comprises a request from the client, one or more provisional responses and one final response. Every SIP request contains a method, request URI and SIP version [1].

1) **Method**: It defines six methods namely:

a) **REGISTER**: It is used by a client to register a particular address with the SIP server.

b) **INVITE**: A session is being requested to be setup using a specific media.

c) **ACK**: Message from client to indicate successful response to an INVITE has been received.

d) **OPTIONS**: A query to a server about its capabilities. This allows a client to discover information about the supported methods, content types etc.

e) **BYE**: A call is being released by either the server/client.

f) **CANCEL**: Cancels any pending requests. It is usually sent to the proxy server to cancel searches.

2) **Request URI**: The Request URI is a SIP URI. It indicates the user or service to which this request is being addressed.

3) **SIP Version**: Applications sending SIP messages must include a SIP-Version of “SIP/2.0”.

D. SIP responses

SIP responses are distinguished from requests by having a Status-Line as their start-line. A Status-Line consists of the protocol version followed by a numeric status code and its associated textual phrase, with each element separated by a single SP (space) character.

Status-Line = SIP-Version SP Status-Code SP Reason-Phrase CRLF

Ex: SIP/2.0 404 NOT FOUND

The Status-Code is a 3-digit integer result code that indicates the outcome of an attempt to understand and satisfy a request. The Reason-Phrase is intended to give a short textual description of the Status-Code.

The first digit of the Status-Code defines the class of response. The last two digits do not have any categorization role. For this reason, any response with a status code between 100 and 199 is referred to as a “1xx response”, any response with a status code between 200 and 299 as a “2xx response”, and so on. SIP/2.0 allows six values for the first digit:

a) **1xx:** Provisional – request received, continuing to process the request.
b) **2xx:** Success – the action was successfully received, understood, and accepted.
c) **3xx:** Redirection – further action needs to be taken in order to complete the request.
d) **4xx:** Client Error – the request contains bad syntax or cannot be fulfilled at this server.
e) **5xx:** Server Error – the server failed to fulfill an apparently valid request.
f) **6xx:** Global Failure – the request cannot be fulfilled at any server.

E. SIP Headers

A header is a component of a SIP message that conveys information about the message. It is structured as a sequence of header fields. Headers provide information about the request (or response) and about the body it contains. The header consists of the header name, followed by a colon and followed by the header value.

V. SESSION DESCRIPTION PROTOCOL

Session description protocol (SDP) is a protocol for describing audio, video and multimedia sessions. Participants are expected to agree on the values, timings, capabilities and desired media formats. The contents of SDP are session name, duration for which the session is active, media comprising the session, owner/originator of the session and how to receive the media (addresses, ports etc). The information that needs to be represented in SDP for T.38 is the fact that T.38 is to be used, whether to use TCP or UDP for transport and which type of error control to be used by T.38.

The default message body type in SIP is application/sdp. The calling party lists the media capabilities that they are willing to receive in SDP in either an INVITE or in an ACK. The called party lists their media capabilities in the

200 OK response to the INVITE. Because SDP was developed with scheduled multicast sessions in mind, many of the fields have little or no meaning in the context of dynamic sessions established using SIP. In order to maintain compatibility with the SDP protocol, however, all required fields are included. A typical SIP use of SDP includes the version, origin, subject, time, connection and one or more media and attribute fields.

VI. PROPOSAL

I propose to develop real time IP faxing using T.38 over SIP on a multi function printer (MFP) to send fax from one internet aware facsimile (IAF) device to another. Real time IP faxing using H.323 exists. The fax channel is set up by Session initiation protocol (SIP). It is an application-layer control protocol to establish, modify and terminate multimedia sessions. In the project it will be mainly used for call signaling and control. SIP is preferred because it is simple and efficient. Currently multi function printers exchange their capabilities like drivers installed, device name by a process known as training. The format supported is tiff. In the system to be developed, connection establishment and release is done using TCP/UDP. SIP is to be used for session establishment and release. Session description protocol (SDP) must be used for describing the capabilities and T.38 must be used for real time IP faxing. It must be ported to work on printers. The fax message received from the sender machine must be printed by the destination printer.

VII. CONCLUSIONS

This paper gives an overview of real time IP fax session and the use of session initiation protocol for session establishment. In order to realize IP fax session in real time, the use of T.38 protocol is essential.

Session Initiation Protocol (SIP) allows the establishment of real time internet fax communications. The T.38 fax call scenarios include various aspects of the call sequence: the detection of fax transmission, the usage of the T.38 session description attributes and the session termination.

The advantages of reduced cost and bandwidth savings of carrying fax over packet networks are associated with some of Quality of Service (QOS) issues like jitter and network hold-ups if the network experiences abnormally high traffic. Though the QoS issues are significant, the future of this approach will be driven by substantial cost savings and applications which are made possible by using this technology.

REFERENCES

[1] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson,R.Sparks,M.Handley,E.Schooler,ColumbiaU./Ericsson/Worldcom/ Neustar/dynamicsoft/ICIR/AT&T, Network Working Group (*rfc3261- SIP: Session Initiation Protocol*), June 2002.

[2] ITU-T Recommendation T.38, Series T: Terminals For Telematic Services, "*Procedures for real-time Group 3 facsimile communication over IP networks – T.38*", April 2007.

[3] ITU-T Recommendation T.30, Series T: Terminals For Telematic Services, "*Procedures for document facsimile transmission in the general switched telephone network – T.30*", July 2003.

[4] http://www.ietf.org/rfc/rfc2327.txt

[5] http://home.howstuffworks.com/foip.html