Building a Secure Short Duration Transaction Network

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by Andrew Gin
University of Canterbury
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Abstract

The objective of this project was to design and test a secure IP-based architecture suitable for short duration transactions. This included the development of a prototype test-bed in which various operating scenarios (such as cryptographic options, various IP-based architectures and fault tolerance) were demonstrated. A solution based on SIP secured with TLS was tested on two IP based architectures. Total time, CPU time and heap usage was measured for each architecture and encryption scheme to examine the viability of such a solution. The results showed that the proposed solution stack was able to complete transactions in reasonable time and was able to recover from transaction processor failure. This research has demonstrated a possible architecture and protocol stack suitable for IP-based transaction networks. The benefits of an IP-based transaction network include reduced operating costs for network providers and clients, as shared IP infrastructure is used, instead of maintaining a separate IP and X.25 network.

Acknowledgements

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Contents

Chapter 1

Introduction

Transaction networks are deployed in many scenarios for example: fire/burglar alarms, remote monitoring/metering systems, airport check in systems and automated teller machines (ATMS). A transaction has the following properties (from [28]):

- Asymmetrical Model: the two endpoints have distinct roles; one is a client which initiates transactions, and the other is a server, which processes requests.
- Half Duplex Transfers: it is unnecessary to send data in both directions simultaneously.
- Short Duration: transactions last between 100s of milliseconds up to several seconds, but never hours.
- Low Delay: latency is usually low.
- Few Data packets: typically two packets, a request and a response.
- Message Orientation: messages, rather than a stream of bytes, are transferred.

Mission critical transactions (such as medical alert systems and ATMs) may have additional characteristics (from [117]):

- High Call Rates: some deployments involve a large number of clients connecting to a smaller number of servers.
- High peak to average load ratio: for example ATMs may average 20 transactions per second for most of the year, but may increase to 200 during the busiest day in December.

One visible application of transaction networks is Electronic Funds Transfer at Point of Sale (EFTPOS). EFTPOS was launched in New Zealand in 1984 and has since become a popular method for making payments. EFTPOS runs over the Telecom Transaction Service (TTS) network. TTS is based on legacy protocols: High-Level Data Link Control in Normal Response Mode (HDLC NRM, also known as Synchronous Data Link Control or SDLC) on the transaction terminal side and X.25 on the bank switch delivery side. The transport layer is AS2805.1, a connectionless datagram protocol. The AS2805 group of protocols is a standard specifically designed for electronic funds transfers. AS2805 encompasses a transport protocol (.1), a financial messaging protocol (.2) and message security, authentication, security and key management (.3-.6) among others. AS2805.2 is an ISO 8583 message format, a specification for financial messages used by payment cards.

TTS uses Public Switched Telephone Network (PSTN) dial up access for sites with low transaction volumes and dedicated leased line access for locations with higher transaction volumes. In addition to transaction terminals and transaction processors, TTS infrastructure is also comprised of concentrators (to aggregate transactions) and Network Access Controllers (NACs). NACs provide protocol conversion, proxying, concentration of transactions, routing and network management functions.

TTS is based on protocols that have been superseded by IP based protocols. IP is ubiquitous; many large organisations have deployed IP networks, while broadband DSL and wireless access bring IP connectivity to small businesses and residential users. Voice telephony is also migrating to IP as Voice over IP (VoIP). Running multiple services over a shared IP infrastructure reduces costs as providers no longer need to maintain and operate a dedicated, specialised network that provides for only a single application. Thus, EFTPOS will eventually need to be migrated to operate over IP.

Transaction terminals each need their own telephone line or leased line. When IP is used, however, multiple terminals can share the same link. Line rental costs can be reduced by connecting terminals to an existing DSL service instead of providing each terminal with its own phone line. Adding a terminal to an existing LAN or DSL service is also faster than adding an additional telephone line at the merchant's premises.

Broadband services are 'always on'. Transaction terminals can take advantage of this as no connection time is required. Broadband DSL speeds are also at least several megabits per second in speed. This is sufficient to support hundreds of transaction terminals in addition to existing uses of DSL such as internet use and VoIP [17].

If the EFTPOS service can operate over IP, it opens up the potential to use any medium which can carry IP, for example wireless broadband, GPRS and CDMA. This creates the possibility for a mobile real time EFTPOS service, for example in taxis and temporary stalls. Current mobile EFTPOS services are batch processed offline (at the end of the day for example).

In addition to business drivers, such as reducing operational costs, for migrating from legacy protocols to modern protocols, technical drivers exist as well. As X.25 is progressively replaced with IP networks, X.25 skills will diminish, for example planning, configuring and maintenance. There is also the potential reduction of vendor support. Furthermore, replacing X.25 infrastructure as hardware becomes obsolete, also becomes more expensive.

The newer protocols also improve on X.25. X.25 provides a robust, error free link between two end points.[1] X.25 buffers the entire frame, checks it for errors, then forwards it to the next node. The cost of this error correction is a high delay. While this has little effect for large transfers, it is noticeable with small exchanges such as an EFTPOS transaction. This buffering behaviour increases the memory required in X.25 hardware, also making it expensive.

X.25 is optimised for noisy links with high error rates. However, newer digital transmission media have improved in quality, making X.25's robust error correction unnecessary. The high latency as a consequence of this error correction also results in poor bandwidth utilisation; small packet and window sizes, in addition to the latency, cause X.25 to lose its effectiveness when the line speed is greater than 100kbps. Modern broadband speeds, even for residential users, are typically much higher than this.

1.1 Research Goals

This research aims to define a generic IP based transaction protocol stack on which to run a transaction oriented application. The current EFTPOS system will be used as a benchmark system.

This project does not intend to define a new EFTPOS protocol, that is, a replacement for AS2805, but rather define a new generic architecture that can carry AS2805 or any other transaction oriented protocol.

The new architecture must at least have the same performance and characteristics as the current benchmark EFTPOS system, but should be capable of exceeding them. These characteristics include security, reliability, availability

[1] `http://www.sangoma.com/main/support/tutorials/x25`

and transaction time.

This requires analysing potential messaging/transport and security protocols which satisfy the requirements. From the candidate protocols, a protocol stack can be assembled which forms the basis of possible architectures. The architecture defined is not limited to an EFTPOS application; it is relevant to any system that utilises short transactions.

Chapter 2

Telecom Transaction Service

This section details an example of a current transaction network, the Telecom Transaction Service (TTS). TTS is used for the EFTPOS system in New Zealand. Transaction terminals connect to transaction processors via Network Access Controllers (NAC), as the topology in Figure 2.1 illustrates [66]. Transaction terminals can send transactions in either front end or back end transaction mode. Terminals in front end switching mode send transactions to processors as designated by the card issuer. Thus, a terminal will send transactions to several processors. Terminals in back end switching mode send all transactions, regardless of card issuer, to a single processor. This transaction processor will then either process the transaction itself or send it to the transaction processor elected by the card issuer.

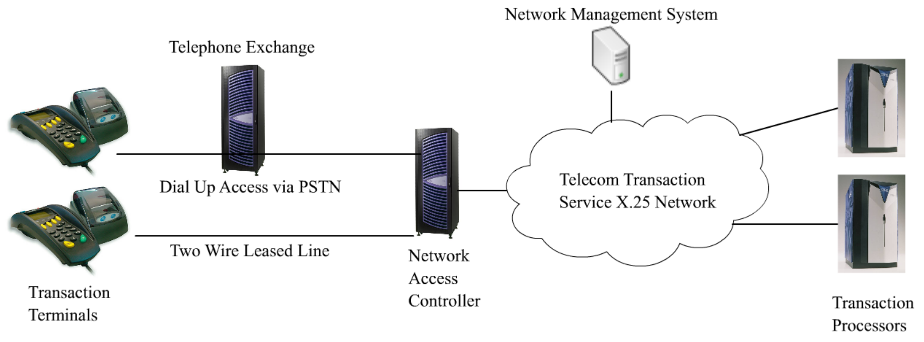


Figure 2.1: The Telecom Transaction Service (TTS) Toplogy.

The protocol stack is in Figure 2.2. Transaction terminals use HDLC(NRM)/SDLC as the Data Link/Network layer protocol to communicate with the NAC. The NAC operates as the NRM primary station and the terminal as the NRM secondary station (that is, the NAC always polls the terminal). The NAC communicates with transaction processors via the TTS X.25 network. The transport protocol AS2805.1 is used end to end from the transaction terminal to the transaction processor.

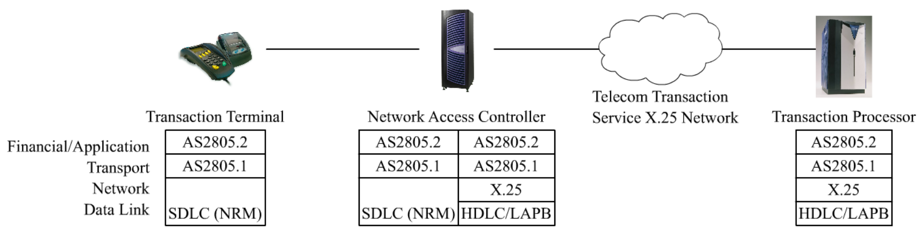


Figure 2.2: The current TTS protocol stack.

2.1 Network Access Controller (NAC)

NACs provide TTS access for transaction terminals. NACs also multiplex, route and accumulate network statistics. NACs concentrate 16 or 32 connections onto a single X.25 connection to the transaction processor. Multiplexing reduces the processing load on the transaction processor, as fewer connections are dealt with. The routing role enables transaction terminals to address different transaction processors as required (explained in Section 2.4). Network statistics accumulated for each terminal and X.25 port can be used by the Network Management System for fault resolution and network planning.

Terminals connect to one of the NAC's 16 or 32 connections in several ways:

Dial Up: This uses the PSTN to access the NAC. This is less reliable and slower than a dedicated connection (the other three methods described below), but allows widely distributed, low transaction volume terminals to be concentrated on a small number of NAC ports. Terminals using this access method are only visible during a transaction, which prevents management of these terminals.

Single Drop: A single terminal can have a dedicated line for continuous access to a single port on the NAC. This is intended for high volume locations which require high reliability and availability. Dedicated lines do not rely on the PSTN; the PSTN can worsen delays and occasionally restrict access.

Multi Drop: Each of the 16 or 32 ports on a NAC can support up to eight dedicated or dial up terminals on a single line. This is intended for multiple terminal sites such as supermarkets and shopping malls.

Terminal Controller: Terminal controllers can address 16 or 32 terminals and are also commonly used in supermarkets, malls and other sites which have a large number of terminals. Terminal controllers appear as a single terminal to the NAC and can be single or multi dropped. This enables multiple transaction terminals to connect transparently to the NAC.

Terminals in the TTS system connect to the NAC through a 1200 bps two wire line (dedicated/leased line or dial up). This line speed is sufficient, as transactions are approximately 300 bytes in length [14]. The latency and transaction times depend on the NAC and transaction processor load, access mode (dedicated or dial up) and line quality. However, one way latency is approximately a few seconds (a mean of less than 1.2 seconds) and total transaction time is approximately 15 seconds [65].

2.2 Network Management System (NMS)

The NMS monitors and controls the TTS network. It allows configuration, software updates, queries, monitoring of NACs and the display of statistics. Queries to obtain status and statistical data (such as traffic volumes and dial attempts) from NACs can be used for fault resolution and network planning. The NMS also gives a degree of service awareness; the NMS periodically polls each NAC to discover its status. NACs also poll the terminals with Receiver Ready messages, requesting transactions; this notifies the terminal it is ready to receive transactions. Conversely, NACs can also send Receive Not Ready messages to terminals indicating the NAC is unable to provide services.

2.3 Security

The TTS X.25 network employs the Closed User Group (CUG) facility to prevent unauthorised access to the network. The network is invisible to Network User Addresses (NUAs are unique identifiers, equivalent to a phone number or an IP) which have not been entered into the list of acceptable NUAs. Different CUGs are also used to establish different security permissions for the different types of traffic (for example management data and transaction traffic) and for billing.

A transaction processor cannot call a NAC or initiate a transaction to a terminal. A transaction processor can only contact the terminal if a circuit already exists, or by calling the NMS which will authenticate its permissions to access the terminal.

Terminals do not connect directly to the transaction processors; terminals must be proxied through a NAC. This reduces load on the transaction processor as they do not see connections and disconnections from terminals, only a stream of transaction data aggregated from NACs. This also improves security as only NACs can connect to transaction processors, not the terminals. This reduces the risk from 'fake' terminals, as they must authenticate with the NAC before they can send transactions.

'Security by obscurity' is used to some degree in this system. Malicious end users are unlikely to have consumer equipment that can interface with SDLC or X.25.

The system is also relatively hard to perform Denial of Service (DoS) attacks against, as all links to NACs are 1200 bps. Links between NACs and transaction processors are 48 kbps, consequently it is impossible for a single terminal to overwhelm a transaction processor.

2.4 Procedure

ISO 7812[1] is a standard for magnetic stripe cards (such as door entry, ATM and credit cards). ISO 7812 defines a Primary Account Number (PAN), which is typically 13, 16 or 19 digits in length. The PAN is normally printed or embossed onto the card itself (Figure 2.3). The first digit is referred to as the Major Industry Identifier (MII). This identifies which industry the card is used in, for example 4 and 5 are banking/financial categories. The first six digits (including the MII) are referred to as the Issuer Identifier Number (IIN). This identifies the card issuer, for example 4xxxxx is Visa, 51xxxx-55xxxx is MasterCard. IINs are assigned by the American Bankers Association and are managed like IPs and radio frequencies.[2] The IIN was also known as the Bank Identification Number (BIN). The next sequence of numbers is the account number. This can be up to 12 digits in length. The final digit is a checksum using the Luhn algorithm.

Inside each transaction terminal is a card prefix table (Figure 2.4). The

[1] `http://www.merriampark.com/anatomycc.htm`

[2] `http://www.ansi.org/other_services/registration_programs/iin_registration.aspx`

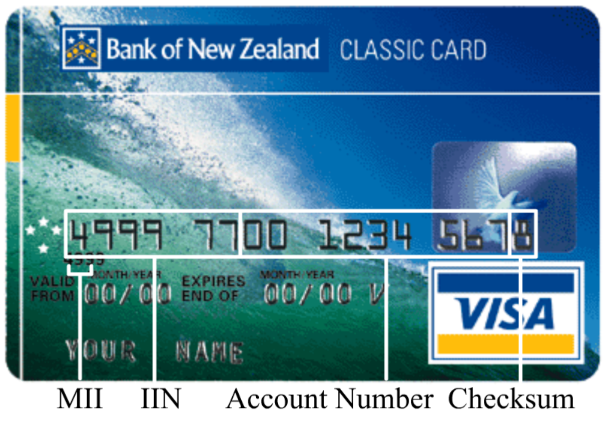


Figure 2.3: The first digit is the Major Industry Identifier (MII). The first six digits (including the MII) is the Issuer Identifier Number (IIN). The seventh to second to last digit is the account number. The final number is a checksum using the Luhn algorithm. Image is from http://www.bnz.co.nz.

terminal uses this table to route transactions from a particular card type to a particular NAC. The first two columns contain the PAN range, for example VISA is 4xxxxx. A row exists for each card type supported by the terminal (for example bank/debit cards, credit cards, petrol cards and customer loyalty cards). The Network International Identifier (NII) is a three digit number used in the TTS by the NACs to map messages from terminals to transaction processors. The Dial Number column is the number of the NAC which relays transactions for that particular PAN range. Other columns also exist, such as a key set and session state.

PAN range		NII	Dial Number	...
40...0	49...9	030	08731	...
...	...	...	...	...

Figure 2.4: The Card Prefix Table embedded within a transaction terminal. An example Visa entry is shown; a row exists for each card type.

The NAC contains an addressing table, mapping NIIs to transaction processors (Figure 2.5). The X.25 numbers for two transaction processors are listed, with the second one used as a back up.

Each transaction terminal is identified by a network address which is built

NII	Transaction Processor 1	Transaction Processor 2
030	4709000	9312000

Figure 2.5: The NAC addressing table. This determines which transaction processor should handle the transaction, based on the NII.

up as it progresses through the network. This address is only unique within a NAC, not between NACs. This address also does not identify which NAC the terminal is connected to. Therefore, transaction processors must use the existing X.25 circuit (created by the NAC to send the initial request) to send replies, or provide its own mapping to establish which NAC the terminal is connected to.

When a card is swiped, the transaction terminal connects to the NAC using the Dial number field in the Card Prefix Table, according to the card type. The terminal uses the NII as the network destination, that is, the transaction processor.

When the NAC receives this, it establishes a virtual circuit to the transaction processor specified in the NII if one does not already exist. A NAC has at most one circuit for each transaction processor. If a virtual circuit cannot be established with the primary transaction processor, the NAC attempts to establish one with the back up secondary address.

After the transaction has been sent, the transaction processor replies (for example an 'Accepted' or 'Declined' response) using the same circuit. After the transaction is completed, the NAC holds the idle circuit up for three minutes before clearing the call. If additional data is transferred (for example further transactions) the timer is reset.

2.5 Previous Approaches

Most available approaches focus on using a legacy gateway to convert protocols [11]. The remote device connects to the gateway (using the legacy protocol) as if it were a telephone exchange. The gateway extracts the application layer data, and sends this to the transaction processor (or any other destination) over the replacement protocol (for example TCP/IP). These legacy gateways, or EFT to IP Converters, can also log transactions to provide statistical information for the merchant.

Several vendors offer this as a solution, such as Hypercom and Braintree Communications. Solutions such as these are a stopgap measure at best, as they essentially enable older equipment to be connected to the new network. These methods will be valuable during a migration phase, when an IP based short transaction network must support both new IP and legacy protocol transaction terminals.

Attempts intended for native IP transaction terminals have also been made. Unfortunately, many of these never progressed past the experimental phase of development, or are proprietary. These are described in detail in Section 3.

Supervisory Control and Data Acquisition (SCADA) systems provide a method to obtain and process data from remote locations. Such systems are typically the foundation of utility control systems [52, 54]. Utility control systems have many applications such as power plants, water/waste utilities, refineries, oil/gas transmission, distribution and production, communication networks and industrial control.

A SCADA system provides monitoring and control of remote devices and processes in real time. Measurements may be pressure, flow rate or temperature and can alert operators to component faults. SCADA systems use a polling scheme similar to the NAC method; the monitoring station periodically polls the remote terminal units (RTU) for monitoring of system statistics. Another polling technique is a round robin method. In this method the monitoring station sequentially polls each RTU. When one RTU has been polled, the monitoring station polls the next RTU. This results in continuous utilisation of the communication line. The sampling rate then depends on the number of RTUs on the line.

While the SCADA architecture has many of the same requirements as a transaction network, for example short delay notification and messaging, the model is slightly different. SCADA monitoring stations initiate connections to the remote devices, while in a transaction network such as EFTPOS, it is the remote devices (transaction terminals) which initiate connections to the central station, the NACs in this case

IBM's Message Queuing Telemetry Transport (MQTT) [19, 85] is another architecture for real time messaging. It has the same goals of remote data acquisition as SCADA, however the model is a subscribe/publish model, as opposed to SCADA's poll/response model. Subscribers (for instance a monitoring station) connect to a repository and subscribe to a topic, for instance flow control in a pump station. The remote device, for example the pump station, publishes messages to the message repository. The repository then distributes the messages to all subscribed clients. The message repository can also store messages for offline clients and deliver them when they come online again. MQTT is based on TCP/IP like third generation SCADA systems [83] and features three Quality of Service (QoS) levels: best effort, at least once and exactly once.

Like the SCADA architecture, MQTT is not aligned with the messaging model of EFTPOS. It would not make sense to publish the data from a single transaction terminal to several subscribers as only a single transaction processor is needed to process a transaction.

2.6 Requirements

Based on the description of the current EFTPOS system, the requirements for the new system can be formulated. The solution has the following requirements:

Low Delay: The current system has a mean round trip latency between the terminal and host of 2.4 seconds (1.2 seconds each way). This depends on the network used. Total transaction time is approximately 15 seconds.

Transaction Aggregation: Aggregating a large number of messages onto one connection reduces costs and network requirements. The reduced connection load on transaction processors also improves performance and reliability. This is currently provided by NACs and terminal controllers.

Reliable Transport: A guaranteed 'at most once' delivery is required, not 'best effort'. The current system uses SDLC and X.25 to provide link level acknowledgement, flow control and error recovery.

Reliable Network with High Availability: The network must have fault tolerance and a degree of 'network intelligence'; terminals will know which servers/connections are down and use alternatives. Terminals also know whether the service is available or not. This requirement includes redundancy, alternate routing and load sharing.

In the current system, terminals using dial up access can determine whether the link is alive by the presence of a dial tone. NACs also poll terminals, notifying them of the NAC status. NACs have a back up transaction processor to establish circuits with, if it cannot establish one with the primary transaction processor. The NMS also periodically polls NACs to learn of their status.

Network Management: With network intelligence, the status of all components in the network should be known at any or all times. This includes reporting, controlling and measuring transactions in the network. Faults must be easily diagnosed and located quickly. This is currently provided by the NMS.

Scalable 1: The system must support a large number of terminals connected to a relatively small number of central processors.

Scalable 2: The system must support high connection rates.

Scalable 3: The system should be designed to support peak, not average load. When the transaction processor is overloaded during congestion, the transaction rate should be throttled.

Secure: Perfect forward secrecy is required with one time session keys. The system must be secure from external risks as well as internal risks (for example rogue or hijacked clients within the network). This is provided in the current system through the use of X.25's CUGs, the proxied access model via the NACs and the specialised equipment required to access the network. The connections to terminals are rate limited, making DoS attacks from a single client impossible.

Efficient 1: The solution must be simple enough to implement easily.

Efficient 2: The solution must operate under limited bandwidth environments. In the current system, terminals use a 1200 bps link. However, with modern networks, speeds of at least several kilobits per second are possible. Modern broadband links are at least several megabits per second.

Efficient 3: The solution must be able to operate on embedded devices with low processing power and perform encryption in reasonable time. The current system only performs encryption at the application level; AS2805 only encrypts the PIN using the Triple DES (3DES) cipher.

Use Active, Industry Supported, Open Standards and Protocols: Open systems and protocols have the benefit of being exposed to peer review, public scrutiny and criticism that in turn may expose performance and security issues. While exposing these makes the protocol potentially more vulnerable to attack, more important is that the vulnerabilities drive development in order to reduce or remove the risk, making the vulnerability a temporary one. This is more desirable than the 'security by obscurity' model, where the security of a system relies on knowledge being kept secret. Knowledge which must be kept secret is a potential point of vulnerability. Open standards also improve connectivity; when proprietary protocols are used, only devices from one particular vendor can be used.

Chapter 3

Candidate Protocols

This section introduces several transport protocols (not necessarily layer 4 in the OSI model) which could support the requirements defined in Section 2.6.

3.1 Transmission Control Protocol (TCP) - RFC 793

Transmission Control Protocol (TCP) is one of the core transport protocols in the Internet Protocol suite, the other being User Datagram Protocol (UDP). It is defined in RFC 793 [97]. As it is widely used and published, it will not be detailed here.

In terms of its suitability for this project, RFC 955 [28] asserts that TCP is at one end of possible transport service attributes and UDP is at the other extreme. TCP provides a reliable flow controlled transfer. Packets can also be fragmented at the sender end and reassembled at the destination. The packet stream offered by TCP means the application must provide its own message framing to distinguish separate messages. TCP also features congestion control features. However, TCP lacks fault tolerance functionality present in newer transport protocols, such as multihoming and link failure in SCTP, described in Section 3.4. Fault tolerance improves reliability as it notifies the client that the network peer is not responding.

TCP features an explicit connection establishment and termination phase. In terms of transactions where the transaction consists of a single packet sent and received in response, the establishment and termination phases would constitute a larger number of packets sent and received than the actual transaction itself. However, this may be a minor issue; when the transaction establishment does make up a large proportion of the total transaction time, the total transaction time is likely to be no more than several seconds. This duration is acceptable for short duration transaction systems such as EFTPOS. This allows a TCP based short transaction system to operate adequately even in an environment with a large RTT.

By itself, TCP cannot be considered a suitable component for this project. While it provides a reliable, flow controlled connection, it is not message oriented and also lacks fault tolerance functionality.

3.2 User Datagram Protocol (UDP) - RFC 768

User Datagram Protocol (UDP) is the other core transport protocol in the Internet Protocol suite. It is defined in RFC 768 [96]. As its operation is widely used and published, it will not be detailed here.

Unlike TCP, UDP is a message based protocol. This relieves the application from having to provide its own message framing. UDP is unreliable, which means packets may be duplicated, lost or unordered. Applications will then need to provide for timeouts and retransmissions.

As UDP is unreliable and does not establish a connection or virtual circuit before sending data, it has no congestion control features. As a result, any congestion in the network may be worsened by UDP. The solution would be to implement congestion control at the application layer, however as there are transport protocols which have congestion and flow control built in, this would unnecessarily complicate the application.

While UDP is a message based protocol and has no connection phase, it lacks reliability and has no flow or congestion control mechanisms. Consequently, UDP by itself cannot be considered a suitable transport protocol for this project. Attempts have been made to build reliability and flow control on top of UDP and one example, QTP, is described in Section 3.5.

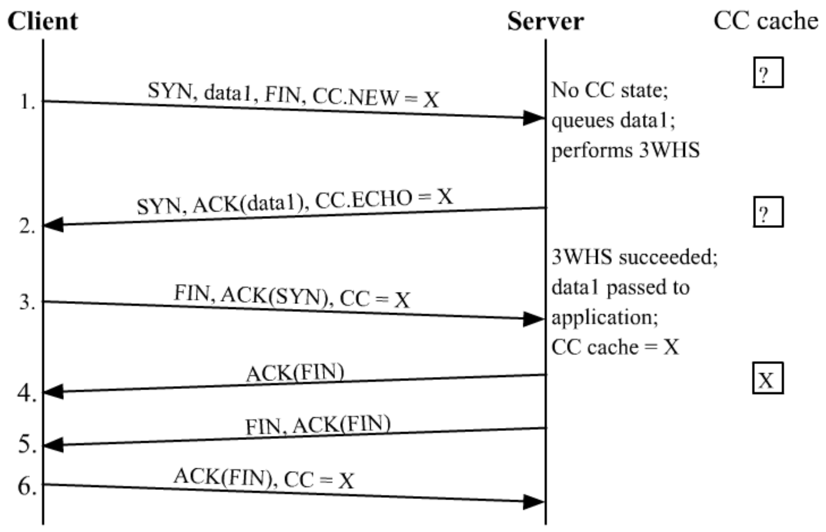
3.3 Transaction TCP (T/TCP) - RFC 1644

Transaction TCP (T/TCP) is an experimental backwards compatible extension to TCP, defined in RFC 1644 [30]. T/TCP is designed to provide reliable and efficient client-server transaction oriented traffic. The extensions aim to make available a transport protocol as reliable as TCP and as fast as UDP. T/TCP improves upon TCP in two ways: by bypassing the 3-way handshake (3WHS) and shortening the delay in the TIME-WAIT state. Bypassing the 3WHS means a T/TCP connection can approach a UDP connection in terms of speed, while maintaining the reliability of a TCP connection [109]. Rather than performing the 3WHS before transferring data, T/TCP carries the user data in the initial connection establishment packet, reducing the number of packets required to perform a transaction to 2, therefore improving transaction times. Shortening the TIME-WAIT delay enables sockets to be cycled and reused more often, increasing the rate at which a client can send transaction requests.

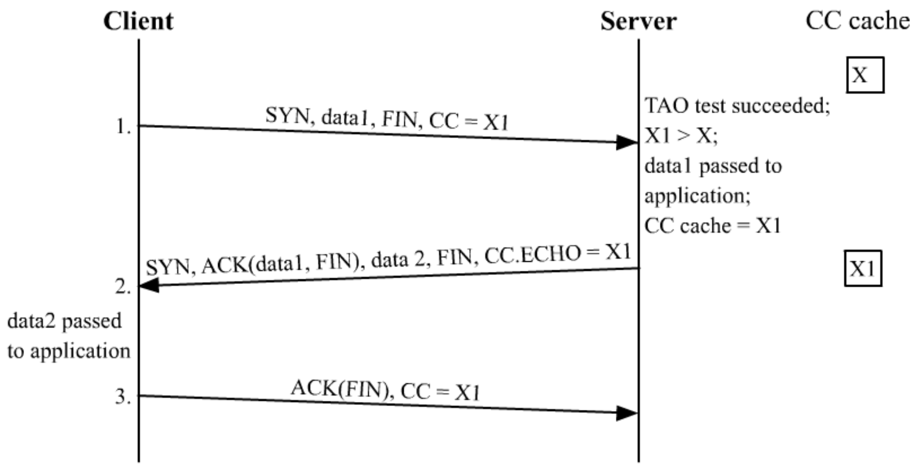
3.3.1 Bypassing the 3-way handshake

The 3WHS is bypassed through use of a mechanism called TCP Accelerated Open (TAO) (Figure 3.1). Standard TCP necessitates the use of the 3WHS to prevent the receiver from confusion by the receipt of old duplicate connection initiations. The receiver of the initial SYN responds with an ACK, verifying with the sender that the connection request is not an old one that arrived out of order. T/TCP achieves this without using the 3WHS by using a monotonically increasing 32-bit incarnation number called a connection count (CC). This is carried in the T/TCP header as a new TCP option anytime a client wishes to initiate a connection with a server. A server supporting T/TCP keeps a cache of the last valid CC for each client. When a client first connects to a server (Figure 3.1(a)), the server has no CC state for that client, so a normal 3WHS is performed. On this initial connection, the server initialises the CC for that particular client on successful completion of the 3WHS. Subsequent connections from that client use an incremented CC value. When the server receives a connection request, the CC value in the connection request is compared with the corresponding client's cached CC value on the server. If the incoming value is larger than the cached value (Figure 3.1(b)), the SYN is considered new and the connection is considered *established*. The user data is passed to the application layer immediately and the cached CC value is updated. If the incoming CC value is not larger than the cached value, a normal 3WHS is performed to validate the connection request, as the server has no way of knowing whether the SYN is an old duplicate, or was delivered out of order. This comparison of CC values is known as the TAO test. If the TAO test succeeds, an optimisation is realised in the form of reducing the connection establishment by one RTT. The user data is passed to the application layer as soon as it is received, otherwise the server queues the user data and falls back to the usual 3WHS of TCP to validate the SYN. Falling back to TCP also provides for reliability and backwards compatibility.

T/TCP effectively reduces the number of segments required to perform a



(a) The first connection attempt requires the use of the 3WHS to validate the SYN.



(b) Subsequent connection bypasses the 3WHS through use of the TAO test.

Figure 3.1: The TCP Accelerated Open (TAO) bypasses the 3WHS to minimise the number of segments exchanged to perform a transaction (adapted from [30] and [111]).

request-response transaction to three segments and because the data is carried in the first two, the applications see the user data at the same speed as if UDP were used. Notice in Figure 3.1(b), the initial segment contains the SYN, data (data1) and FIN. After the data is passed to the application layer, the FIN is processed and the server responds with its SYN, ACK and FIN flags as well as any response data (data2). The client processes the received SYN and ACK, and passes the server's reply to the application layer. The client also uses the CC.ECHO option to validate the server's SYN ACK segment. When the FIN is processed, the client responds with a final ACK and moves into the TIME-WAIT state. When the server receives the ACK of its FIN, the connection is closed. This example shows how a connection is established and closed with a minimal exchange of segments.

It is possible for the CC values to wrap around, for example: connections which last longer than the Maximum Segment Lifetime (MSL), as the CC values are global to all connections; the client crashing, rebooting and reinitialising the CC generator back to 1; or simply reaching the end value. Since all subsequent connections will have lower CC values than the cached value on the server, there will be a performance degradation, as the TAO test will fail. This causes the hosts to perform the 3WHS each time they initiate a new connection. Clients avoid this by specifying the CC.NEW option, which updates the server's CC cache to the new value.

3.3.2 Truncation of the TIME_WAIT state

The TIME-WAIT state exists in TCP to ensure the remote host has received the acknowledgement (ACK) of its connection termination request (FIN). Upon receiving a connection termination request, a host enters the TIME-WAIT state for twice the MSL, usually between 60 and 240 seconds (Figure 3.2). In this state, the port pair (C:P and S:P) cannot be reused; a host can only receive and acknowledge the retransmission of a remote FIN. This keeps old packets from the closed connection still in the network from interfering with another connection created using the same socket pair, should they be reused. For example if a server retransmitted a FIN (as the client's final ACK was lost), the TIME-WAIT state would ensure that the client responded by retransmitting the lost ACK, rather than an RST which would happen if the client was not in the TIME-WAIT state. The wait time of 2MSL also ensures that any remaining packets from the previously closed connection expire.

Shortening the TIME-WAIT state can increase transaction rates and is possible in T/TCP due to the introduction of the CC value. The CC value protects against old duplicates, as the value increases with each new connection. In T/TCP, the time spent in the TIME-WAIT state is a multiple of the retransmission timeout (RTO, of 1.5 seconds [111]) instead of the MSL; it is reduced to 12 seconds, eight times the RTO (Figure 3.3). The RTO multiplied by eight ensures that the sender has an opportunity to retransmit any unacknowledged segments. It is clear, when comparing TCP's TIME-WAIT state (Figure 3.2)

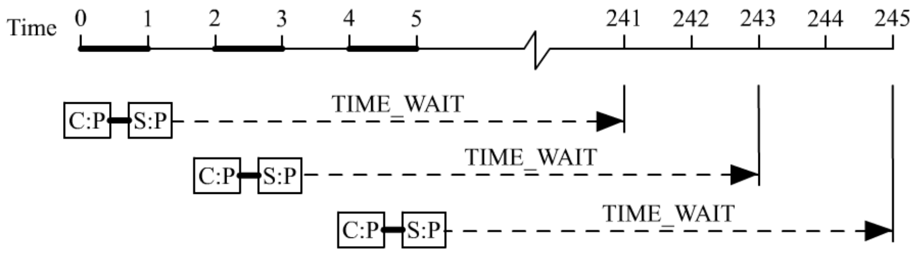


Figure 3.2: TIME_WAIT state in TCP (from [111]). A client-server port pair (C:P and S:P) cannot be reused for 2MSL (typically 240 seconds) after a connection has been closed.

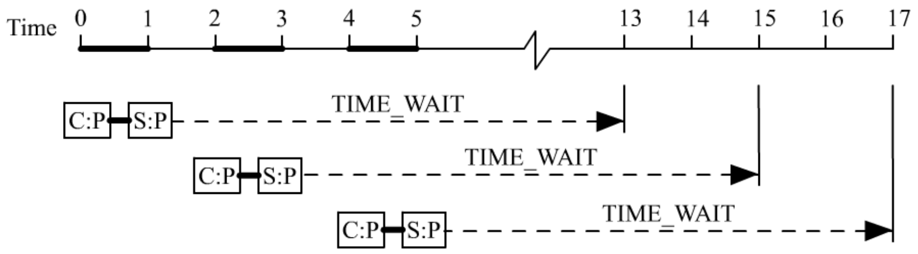


Figure 3.3: TIME_WAIT state in T/TCP (from [111]) when different ports are used for each transaction. The client-server port pair (C:P and S:P) can be used sooner than in TCP.

with T/TCP's (Figure 3.3) that shortening the TIME-WAIT state makes ports available for use sooner and more often.

A further optimisation is possible if the client and server reused the same ports for a subsequent transaction (Figure 3.4). The TIME-WAIT state ends as soon as another transaction is initiated via use of the CC values. For example if a client had received a FIN, an ACK would be sent in response. If this ACK was lost but the client initiated a new transaction before the retransmitted FIN arrived, the exchange would continue as normal. This is possible because the new SYN would have a higher CC value and the TAO test would succeed. This implicitly acknowledges the server's unacknowledged FIN. The server then closes the old connection and starts a new one. The data contained in the newly received SYN is then passed to the application layer (as the TAO test succeeded).

A constraint on the truncation of the TIME-WAIT state is that the duration of the connection must be less than the MSL (which is typical of a simple request-

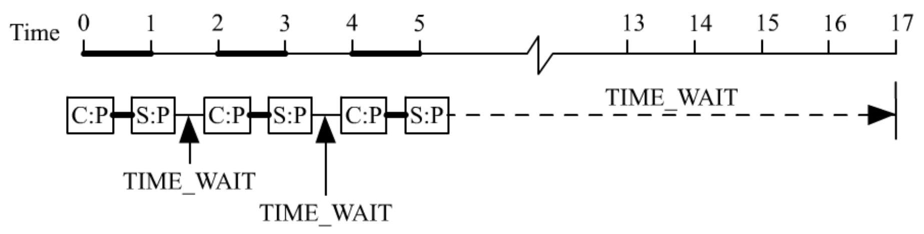


Figure 3.4: TIME_WAIT state in T/TCP (from [111]) when the same ports are reused for each transaction. The client-server port pair (C:P and S:P) can be reused when a new connection is initiated.

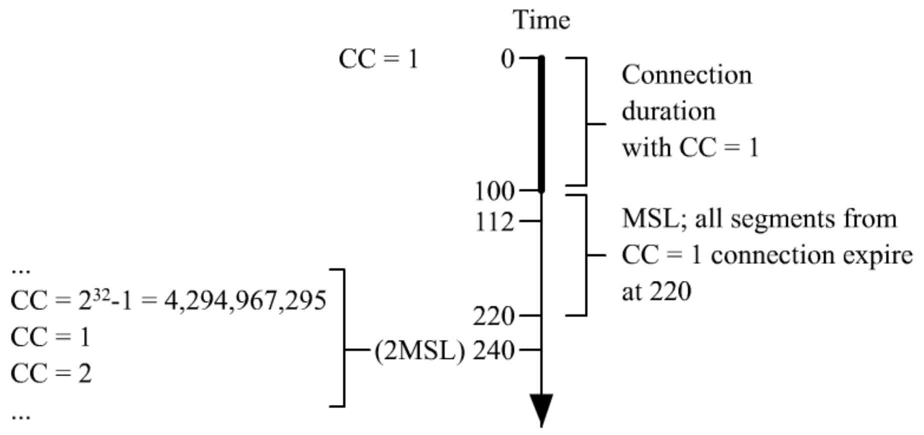

Time
CC = 1
0
Connection duration with CC = 1
100
112
MSL; all segments from CC = 1 connection expire at 220
...
CC = 2^32-1 = 4,294,967,295
CC = 1
CC = 2
...
220
(2MSL) 240

Figure 3.5: Truncation is possible here, as the connection duration is less than the MSL of 120 (from [111]).

response transaction in any case). The T/TCP RFC states that "The essential requirement for correctness of T/TCP is this: CC values must advance at a rate slower than 2^{31} counts per 2MSL where MSL denotes the maximum segment lifetime in the Internet". This requirement ensures that a particular CC value for a particular connection (that is, a port pair) is not reused for at least 2MSL. Thus if the duration of a connection exceeds the MSL (120 seconds in TCP's specification, RFC 793, [97]), the TIME-WAIT state interval must revert to TCP's delay of 2MSL (240 seconds) to protect from old duplicates. Figures 3.5 and 3.6 (from [111]) illustrate why. In Figure 3.5, a connection starts at time 0 with a CC of 1, and lasts for 100 seconds. The TIME-WAIT state starts at time 100 and lasts until time 112 or until the client initiates another transaction using the same ports, whichever comes first. As the MSL is 120, all segments from this connection will have expired by time 220. As the CC values repeat only every 240 seconds (2MSL), in this instance it is safe to truncate the TIME-WAIT state (as the connection's duration was less than MSL).

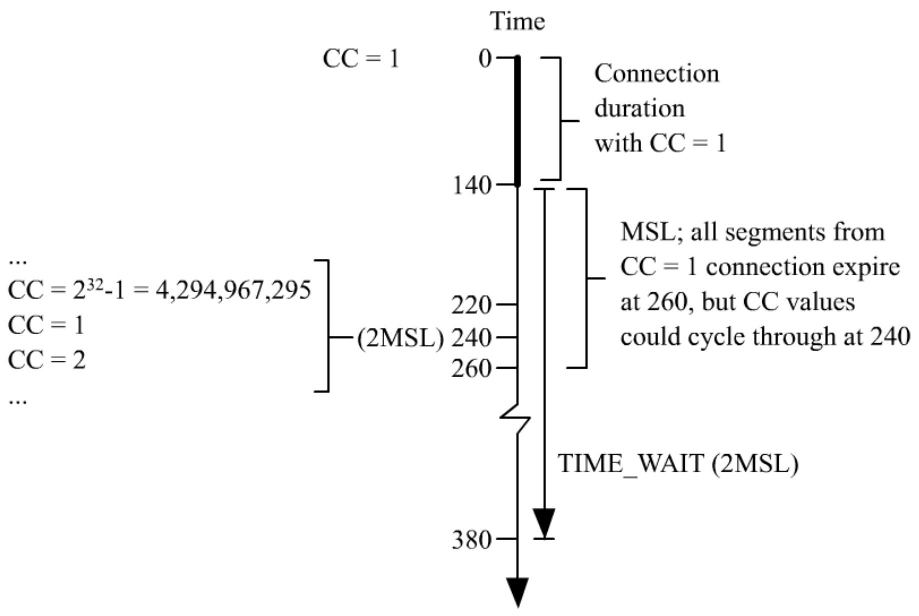

Time
CC = 1
0
Connection duration with CC = 1
140
MSL; all segments from CC = 1 connection expire at 260, but CC values could cycle through at 240
...
CC = 2^32-1 = 4,294,967,295
CC = 1
CC = 2
...
(2MSL)
220
240
260
TIME_WAIT (2MSL)
380

Figure 3.6: Truncation is not possible here, as the connection duration is greater than the MSL of 120 (from [111]); the CC values could wrap around before segments from the first connection expire.

However, in Figure 3.6, the connection lasts for 140 seconds - longer than the MSL of 120. All segments from this connection will have expired by time 260, however CC values cycle through at most every 240 seconds. By time 240, the CC values may have cycled through and a new connection with a CC value of 1 may be initiated. However, segments from the first connection (also with a CC value of 1) are not guaranteed to expire until time 260. Consequently, old duplicates from the previous connection could be delivered erroneously to the new connection and accepted as legitimate, as the CC value is 'correct'. To protect against this, when the connection duration is longer than MSL, the TIME-WAIT state cannot be truncated, and the socket pair cannot be reused for 2MSL (until time 380). The CC value of 1 can be safely reused at time 240 as long as it is for another connection (that is, a different socket pair), however it cannot be reused for the socket pair for which old duplicate segments still exist.

Thus there are two port strategies available: use the same port pairs for every transaction to save TCP resources (Figure 3.4), in which case if the duration of the connection is greater than MSL, the TIME-WAIT state delay must be at least 2MSL, otherwise the TIME-WAIT state can end upon initiation of a transaction; or use different ports for each transaction (Figure 3.3), which means applications do not need to be programmed to use the same port number. In any case, as long as the connection duration is less than the MSL, the TIME-WAIT state is always truncated from 2MSL to 8RTO.

As most transaction oriented connections usually exist for much shorter than 120 seconds, T/TCP offers an optimisation, as resources are available more often and are cycled through faster, making higher transaction rates possible.

3.3.3 Security Vulnerabilities

While T/TCP has attractive features aimed specifically at improving request-response transactions, there are weaknesses which make it more vulnerable than TCP. Spoofing a connection is almost certain to be successful and significantly easier under T/TCP than TCP [46]. When the TAO test succeeds, the server considers the connection to be established and the data is passed to the application layer immediately. Thus, all an attacker needs to do is succeed the TAO test. There are two simple methods to succeed the TAO test, and therefore spoofing a T/TCP connection; the attacker can either perform packet sniffing to discover the current CC value, or simply use a large value for the CC.

If the attacker is on the same network as the host to be spoofed, the current CC value of the connection between the host and the server can be discovered via packet sniffing. Once this is discovered, all the attacker needs to do is increment the discovered CC value and initiate a connection to the server, using the source address of the spoofed host. The monotonically increasing property of the CC value ensures the TAO test will succeed, and the data will be passed to the application layer on the server immediately.

The other method is to simply use a large value for the CC value. If the attacker is unable to sniff out the current CC value, using a large CC value has a high chance of succeeding the TAO test, as all observed implementations initialise the CC value to 1 [105]. The CC value is a 32-bit unsigned number, with a maximum value of $2^{32} - 1$. The chance that a particular host has used even half of these values is unlikely. Therefore the higher the CC value used, the greater the probability of success.

These two methods not only have a high or certain chance of success, they also ensure that subsequent connection requests from the legitimate host fail the TAO test, as the server updates the CC cache with each successful connection. The payload of such spoofed connections can be the same as those used under TCP, such as the command ‘`echo ‘‘+ +’’ >>  /.rhosts`’, used in Unix to extend trust to any user from any host. Sending a spoofed packet with an acceptable CC value to succeed the TAO test, with the Unix command as the payload, is all that is needed to compromise the server. Unlike TCP, sequence numbers do not need to be predicted to establish the connection, making the attack almost effortless, as only one packet is needed. The 3WHS provides a limited degree of validation of the sender’s address, and removing this greatly eases the spoofing of complete connections. Use of the TAO mechanism also negates the use of randomised initial sequence numbers (ISN) for each connection as well as incrementing the sequence numbers every half second - the best techniques for preventing spoofed TCP connections.

Another weakness of T/TCP is its vulnerability to SYN flooding, which leaves the server unable to accept new connections, as all the resources are devoted to multiple half-open connections due to spoofed connection requests. While TCP is also vulnerable, under T/TCP, SYN flooding is more harmful and harder to defend against. To reduce transaction overheads, T/TCP carries data (a request from the client) in the initial SYN. If the TAO test fails, the server queues the data until the fallback 3WHS succeeds. As a consequence, a SYN flood can do more damage under T/TCP, as an attacker can spoof SYNs such that each connection request fails the TAO test. With data queued from many connection requests (due to each one failing the TAO test) memory buffers can be exhausted faster than if TCP were used. The reduced TIME-WAIT state also enables attackers to send requests faster when attacking from the same address and port, which makes T/TCP very prone to DoS attacks.

Strictly speaking, TCP is also vulnerable to queuing the data of pending connections. TCP, however, can be implemented such that data sent with the initial SYN can be discarded. This would require the client to retransmit the data, after the connection is established with the 3WHS. However, if this were applied to T/TCP (which is completely feasible) it would not be aligned with the goal of providing fast transaction processing as the client would need to retransmit the data, effectively reducing the speed to that of TCP.

A method to protect against SYN flooding is the use of SYN cookies [118]. A SYN-cookie is a cryptographically generated ISN, based on the source IP address, port and other data. When a SYN is received to request a connection, the server responds with a SYN-ACK, with the SYN-cookie as the ISN. When an ACK is received in response, the sequence number in the ACK is validated, and only then does the server allocate state, a buffer and open the connection. If the sequence number is invalid, the packet is dropped. Use of the SYN-cookie makes a stateless handshake possible, as resources are allocated only on receipt of the final ACK from the client, rather than on sending the SYN-ACK. As a result, the risk from SYN flooding is greatly reduced, as half-open pending connections cannot exist. Unfortunately this technique cannot be used with T/TCP. The TAO makes use of the CC value, which is carried in the TCP options part of the header. As the server does not hold any state for a connection until the final ACK form the client is received, any TCP options in the initial SYN (and therefore the CC value) would be lost, meaning the CC value cannot be cached.

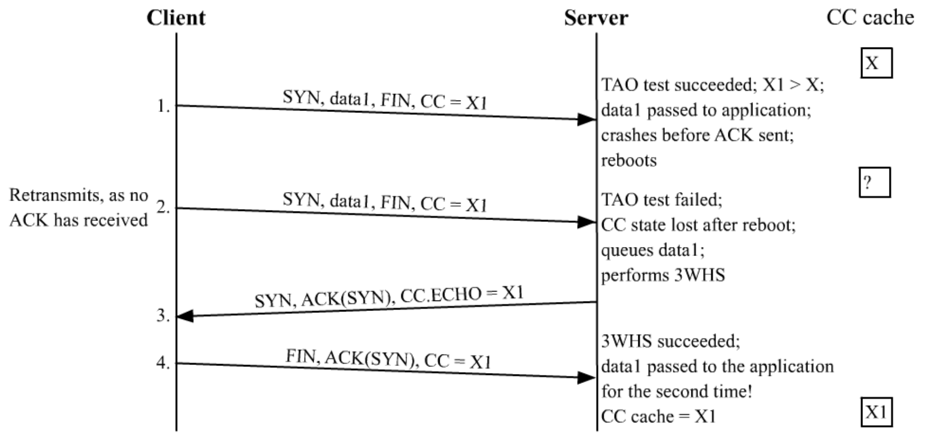

Client
Server
CC cache
X
1.
SYN, data1, FIN, CC = X1
TAO test succeeded; X1 > X; data1 passed to application; crashes before ACK sent; reboots
?
Retransmits, as no ACK has received
2.
SYN, data1, FIN, CC = X1
TAO test failed; CC state lost after reboot; queues data1; performs 3WHS
SYN, ACK(SYN), CC.ECHO = X1
3.
3WHS succeeded; data1 passed to the application for the second time! CC cache = X1
4.
FIN, ACK(SYN), CC = X1
X1

Figure 3.7: T/TCP can deliver duplicate data due to use of the TAO mechanism (from [105]).

Finally, T/TCP is not completely compliant with TCP in some instances. Use of the TAO mechanism can cause duplicate data to be delivered 'legitimately' to the application layer. This is illustrated in Figure 3.7. In this example (from [105]), the client sends the first part of a transaction (the request) to the server. The two hosts have established their CC values, so the TAO test succeeds. The server passes the data in the request to the application layer, however it crashes before it can send its ACK (the response part of the transaction). The client times out as it has not received an ACK, so retransmits the request. After the server has rebooted, it receives the request again, however this time the TAO test fails, as the server's CC cache is reinitialised and thus invalid. Consequently, the server queues the data carried in the request, and performs the 3WHS. When this is completed, the queued data is passed to the application layer for the second time as the server maintains no state after a reboot, and therefore cannot identify duplicate requests. A required attribute of a transaction system (as described in Section 2.6) is 'exactly-once' semantics, and the functional specification of T/TCP does not conform to this.

In addition, page 73 in RFC 793 states "Do not process the FIN if the state is CLOSED, LISTEN or SYN-SENT since the SEG.SEQ cannot be validated; drop the segment and return". This means when a standard TCP server receives data with both the SYN and FIN flags set (that is, from a T/TCP client), the segment will be dropped; sending a FIN with a SYN flag violates TCP's processing rules [58]. Most servers drop the FIN and processing will require a 3WHS to continue, however, T/TCP will be unable to communicate with TCP servers that conform strictly to the TCP specification. In this situation, T/TCP is not compatible with TCP.

3.3.4 Implementations and Development

RFC 1644 was published in 1994, and remains in experimental state. There have been several implementations of T/TCP in several operating system kernels: Linux, SunOS and FreeBSD.[1] Patches exist for up to Linux kernel version 2.4 (which was released in January 2001), however the development[2] never went past the beta stage. Backward compatibility problems also exist in Solaris implementations, when a T/TCP enabled client connects to a standard TCP server. In Solaris 2.4 (SunOS 5.4), a standard TCP server's stack dropped the data in the initial SYN and did not acknowledge it [111]. This caused the T/TCP client to timeout and retransmit the data. A patch was issued in 1998[3] for the FreeBSD kernel, addressing the spoofed connection vulnerability (when the .rhosts file is used as the only method of authentication).

Enhanced Transaction TCP [27] proposes solutions for the security and backward compatibility problems in the current experimental specification. To reduce the risk from spoofed connections, CC value is made a socket variable, instead of a global variable as defined in RFC 1644. This has two advantages; the server can now handle up to $2^{32} - 1/2MSL$ transactions per second *per client*, instead of $2^{32} - 1/2MSL$ transactions per second globally. Secondly, the CC values can now have predictable increments as the CC value spaces are now specific to each client. Consequently the TAO test can be modified so CC values must be exactly one larger than the cached value. Another measure suggested was to randomise the CC value instead of initialising it to 1. This would reduce

[1] `http://www.kohala.com/start/T/TCP.html`. This site also lists several T/TCP enabled internet hosts.

[2] `http://sourceforge.net/projects/ttcplinux` (accessed February 2007).

[3] `http://ciac.org/ciac/bulletins/i-051.shtml`

the ease of spoofing a connection, which only needed to use a high enough CC value to succeed.

To reduce the risk from SYN-flooding, use of a SYN-cache was suggested in [79]. Normally when a server receives a SYN, the whole Transmission Control Block (TCB)[4] for that connection is allocated. Thus when a SYN flood is taking place, the server's resources are tied up in the TCB's of the attacker's pending connections. SYN caching, on the other hand, allocates minimal state when the server receives a SYN. This state records TCP options which are not retransmitted in the ACK from the client, such as the CC value. The full TCB is allocated only when the connection is established. A limit is placed on the amount of cached state to place a bound on the amount of memory the SYN cache uses. When this limit is reached, the oldest entry is dropped. Use of the SYN cache therefore limits the damage potential from a SYN flood. In addition, following the failure of the TAO test, the server should not queue the data, as this can aid an attacker when performing a DoS attack by consuming more memory.

As for duplicate deliveries, two-phase commits and transaction logging can eliminate this problem, that is, make the application layer responsible [109].

3.3.5 Summary

T/TCP was designed for simple request-response transactions. It aimed to provide a protocol as fast as UDP, but with the reliability of TCP. It reduced the overhead required to establish a connection by including the data in the initial connection establishment segments and by bypassing the 3WHS through use of the CC value. This enabled the application layer to receive the data as fast as if UDP were used. The other main improvement upon TCP was the truncation of the TIME-WAIT state. This reduced the time ports were unavailable for use after closing a connection, which enabled higher transaction rates. Unfortunately the use of the CC value greatly eased the spoofing of connections, as all attackers needed to do to ensure success was to use a large enough CC value - a CC value larger than the cached value on the server was all that was needed to accept the data and pass it to the application layer. With TCP, the attacker had to at least guess the sequence numbers and this could be made more difficult with randomised sequence numbers and half-second increments to the numbers - defence techniques that cannot be used to protect T/TCP. The other major security flaw in T/TCP was its risk from SYN flooding; while TCP is also vulnerable, T/TCP would typically exhaust the server's memory faster. The most

[4]An operating system uses these to maintain a transport protocol's connection.

effective protection against this, SYN cookies, again cannot be used to protect T/TCP. In some instances, T/TCP is not compatible with TCP, leading to duplicate message delivery to the application layer, or no communication at all with a strictly implemented TCP host.

There have been limited deployments and implementations of T/TCP and a distinct lack of active development in the protocol. Enhanced T/TCP proposes solutions to T/TCP's shortcomings, however it has not had any deployment or implementation in any major Unix/Linux distribution. The requirements that T/TCP had set out fulfil are highly aligned with those in Section 2.6. In fact, parts of those requirements in Section 2.6 are adapted from RFC 1644, and T/TCP in concept, meets those requirements. However, given the serious security issues, its lack of development, implementation, deployment and maturity, it cannot be considered a viable transport protocol suitable for the requirements described in Section 2.6.

3.4 Stream Control Transmission Protocol (SCTP) - RFC 2960

Stream Control Transmission Protocol (SCTP) is a transport layer protocol originally designed to transport SS7 telephony messages over an IP network [53]. SCTP was developed by the IETF Signalling Transport (SIGTRANS) working group and is currently a proposed standard as RFC 2960 [114].

The information carried in SS7 PSTN signalling messages are for call management, such as call set up and teardown, billing and routing and value added services, for instance call waiting and caller identification [32]. Such messages require reliable and timely delivery as errors or delays can result in disruption of the service (for example failure to establish a call or billing errors). With the growth of VoIP, a transport protocol was needed to transport SS7 signalling messages over IP. Until late 2000, TCP and UDP were the only prospective transport layer protocols in the TCP/IP suite. UDP does not provide reliable, in-order transport, or any form of congestion control. Without flow control, the send rate remains constant, worsening the effects of any network congestion. Congestion in turn makes UDP even more unreliable as data will be discarded. Because of these shortcomings, UDP was not considered. TCP, with respect to carrying PSTN signalling, had the following limitations (from [114]).

- TCP provides strict order-of-transmission delivery. Applications that do not require sequential delivery may suffer from unnecessary delays due to

Head-of-Line (HoL) Blocking

- As TCP is stream-oriented (or byte-oriented), it delivers a stream of bytes. Applications that process individual messages (such as a PSTN signalling message) need to add their own application level message framing within the TCP byte stream. Applications must also explicitly use TCP's PUSH facility to ensure complete messages are sent with minimum delay (as opposed to the standard behaviour of blocking data together and sending when convenient).
- TCP does not support multihoming; a single destination IP address is bound to a particular TCP connection. If this connection fails, it must be re-established. This does not meet the signalling transport's requirement of high availability and network resilience.
- TCP is relatively vulnerable to DoS attacks, such as SYN-flooding

As a result of TCP's limitations, SCTP was proposed to carry PSTN signalling across an IP network. Even though SCTP's design was motivated by this need, it is considered a general-purpose protocol; like TCP, SCTP provides a reliable, full duplex connection with congestion avoidance mechanisms.

SCTP's congestion control functions are based on the rate-adaptive, window-based scheme of TCP. This includes slow start, congestion avoidance and fast retransmit [113]. As these are derived from those of TCP, SCTP is 'TCP friendly'; SCTP applications can co-exist and share network resources with TCP applications without constrained or excessive use of the network. However, SCTP improves upon TCP's congestion window (`cwnd`). A congestion window reduces the sending rate during network congestion by limiting the number of bytes that can be inserted into a network. When no congestion is detected and all the data in a window is acknowledged by the receiver, the slow start and congestion avoidance algorithms (defined in RFC 2581, [21]) double the value of `cwnd`. This is because the sender's current sending rate does not appear to be causing congestion. These algorithms assume that the `cwnd` is limiting the sending rate. However, an application can send at a rate slower than what the `cwnd` is limited to. This is common for signalling applications, as this ensures the network remains functional in the event of a load spike [33]. Under this situation, TCP would increase `cwnd` exponentially. If the application has a sudden burst of traffic, the large `cwnd` will allow all of it to be sent at once. This could cause network congestion resulting in packet losses and timeouts. In SCTP, `cwnd` can only be increased when the full `cwnd` is used, rather than doubling it upon successful transmission as in TCP [53].

SCTP's flow control is also derived from TCP, more specifically TCP Selective Acknowledgements (SACK). TCP SACK is defined in RFC 2018 [81]. The ACK in TCP is used to detect packet loss and is cumulative; it indicates the last packet successfully received in sequence (therefore implicitly acknowledging that all packets before the acknowledged packet were also received successfully). SACK enables the receiver to acknowledge packets that have been received successfully after packet loss. For example (from [33]) if a host receives packets 1, 2, 3, 5 and 7, it will respond with ACK(1), ACK(2), ACK(3), ACK(3), ACK(3). When the sender receives ACK(3), it knows that packets up to 3 were successfully received in sequence. However, receiving ACK(3) multiple times indicate that packet 4 was lost and needs to be retransmitted. The sender, however, is unaware which packets after 3 were also lost or successfully received, so packets after 4 are also retransmitted. With the SACK extension, the receiver would have sent ACK(3)-SACK(5,7). The sender would then know to not only retransmit packet 4, but also packet 6. This reduces the impact of multiple dropped segments, as only the packets that were not sent successfully are retransmitted thus saving bandwidth. Like congestion control, while SCTP's flow control is based on TCP's, SCTP improves upon TCP's mechanism - while SACK is optional in TCP, it is a standard feature of SCTP.

Common Header
Data Chunk 1
. . .
Data Chunk n

Figure 3.8: The SCTP packet format. Multiple data chunks can be in a single SCTP packet up to the MTU.

While SCTP inherits some features from TCP and can offer the same functionality, it contains additional features to overcome TCP's limitations. One of these is conservation of message boundaries. Like UDP, SCTP is message oriented. This simplifies applications that process individual messages, as the transport protocol indicates the start and end of the message. One of the disadvantages of TCP is that the application must preserve the message boundaries themselves. SCTP frames messages by carrying them in units called *chunks*. A chunk consists of a header and the data. As illustrated in Figure 3.8, an SCTP packet can contain multiple chunks. Large messages can also be fragmented into multiple chunks and then reassembled at the receiving end. Fragmenting user messages at the transport layer avoids fragmentation at the IP layer. IP layer fragmentation is undesirable as the protocol header (TCP, SCTP, UDP etc), containing the destination and source port numbers, is contained only in the first fragment [33]. As subsequent fragments do not contain this header data, they may not be able to pass through NAT devices or firewalls. If the path MTU changes (due to a change in the network path for example), SCTP in some circumstances may be forced to perform IP fragmentation. If the receiver is behind a NAT device, the data may never be received, causing the SCTP *association* (equivalent to a connection in TCP) to timeout. This limitation is described in [33].

Another important feature is multihoming. SCTP allows each endpoint of an SCTP association to be accessible from more than one IP address. This provides a level of redundancy (as required for SS7 signalling transport), as the failure of the primary address (for example due to link failure or congestion) does not disrupt the data transfer. When SCTP initialises as association, each endpoint exchanges a list of addresses or host names from which they can receive data from. The sending host selects one of the receiver's addresses as the primary address and uses this for all data transmission. Retransmitted data is sent to an alternate address to improve the probability of successful transmission [87]. Sustained failure to send to the primary address causes the end point to send all data to the alternate address, until the primary address becomes available again. This change is potentially transparent to the sending application [71]. If this occurred in TCP, the connection would collapse, interrupting the data transfer, as the connection would need to be re-established.

A requirement for multihoming to be effective is that the multiple addresses need to be connected to different networks (for example different network address prefixes or possibly ISPs) to ensure the traffic is transmitted over a different network path in the event of primary address failure. The more diverse the alternate network paths, the more fault tolerance multihoming affords; multihoming loses some effectiveness when the alternate paths meet at a single point of failure, such as a single link or router.

SCTP detects the failure of a link via ACKs received from acknowledged data and a heartbeat mechanism. A heartbeat request is sent periodically to each of the remote endpoint's idle alternate addresses to determine the state of the link. The remote endpoint responds with a heartbeat acknowledgement, to confirm the link and endpoint's availability. When the number of unacknowledged retransmissions or heartbeat requests exceed a configured maximum parameter (RFC 2960 recommends a value of 5 for the Path.Max.Retrans parameter), the address is deemed to be unreachable, and an alternate address is selected to transmit data. Heartbeat requests are still sent to unreachable addresses, so that addresses that do become reachable again can respond with a heartbeat acknowledgement, and can be noted as active.

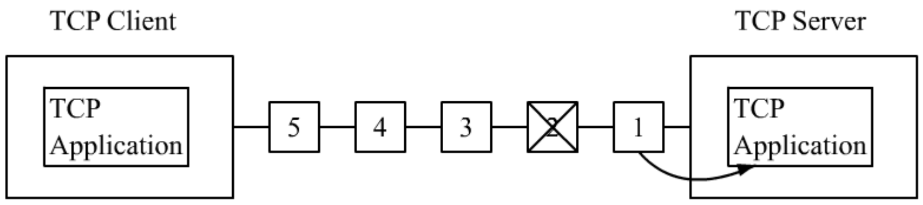


Figure 3.9: Head of Line blocking in TCP. Even though segments 3 to 5 are transmitted successfully, they cannot be delivered to the application until segment 2 is successfully transmitted.

SCTP also adds multistreaming. An endpoint can only have one SCTP association with a particular endpoint (an endpoint can have many concurrent associations as long as they are with different endpoints [115]). Each association, however, can contain multiple streams. Each stream is a unidirectional flow of chunks; each chunk belongs to one stream. Streams are independent of each other and in this way SCTP separates message reliability and message ordering. This enables endpoints to transfer multiple ordered sequences of messages reliably, without the sequences interfering with each other. This avoids the Head-of-Line blocking problem in TCP, which happens when separate streams of data are multiplexed over a single TCP connection. For example in Figure 3.9, segments 1 to 5 belong to separate independent sessions. Segment 1, 3, 4 and 5 are delivered successfully to the host, however, segment 2 is lost. Segment 1 is delivered to the application, but 3, 4 and 5 are queued as TCP requires order-of-transmission delivery. Thus the delivery of 3,4 and 5 to the application must wait until 2 has been successfully retransmitted and received, even through they belong to different logical sessions. In SCTP, each independent session can be transferred over a separate stream, ensuring that the loss in one stream does not affect the other streams (Figure 3.10), thus avoiding HoL blocking and associated delays.

It is important to note that within a stream, messages are normally still delivered in order-of-transmission, so HoL blocking still affects individual streams. In Figure 3.10, if S1 and S2 were each a stream, stream S2 would operate normally, however stream S1 would be blocked until segment 2 of stream S1 was successfully retransmitted and received by the remote endpoint.

SCTP also supports unordered reliable message delivery. This also avoids

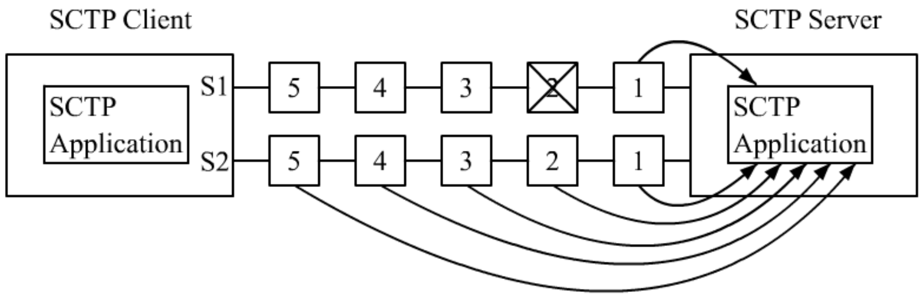


Figure 3.10: Use of multiple streams avoids Head of Line blocking. Even though segment 2 of stream S1 has been lost, segments in stream S2 can still be delivered to the application.

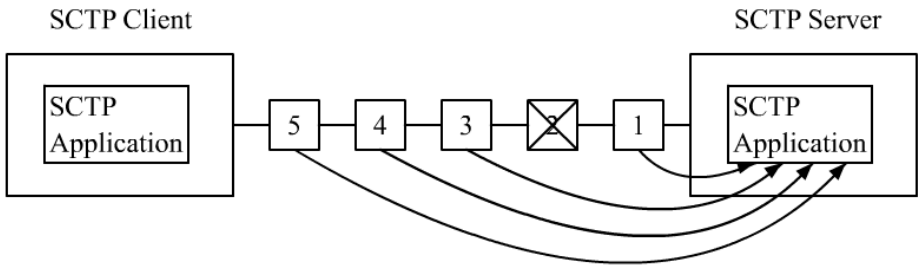


Figure 3.11: Use of unordered delivery also avoids Head of Line blocking. Chunks that have been transmitted successfully can be delivered immediately to the application, regardless of previously lost chunks.

HoL blocking, but this time *within* a stream. For example Figure 3.11 shows a single stream. Chunks 1, 3, 4 and 5 have been specified for unordered delivery, and can therefore be delivered immediately to the application. The delivery mode is specified at the chunk level, so within a single stream, some chunks can be specified for out of order delivery, and the rest delivered in order (the default).

Previous work ([33, 55]) studying the effects of HoL blocking found that the improvements in SCTP over TCP do not result in a large reduction in transmission delay under normal conditions. Both studies found under normal network conditions, the mean delays with SCTP and TCP cannot be statistically differentiated. However, both also found as network conditions worsened (due to congestion and high packet loss), SCTP and its avoidance of HoL blocking provided a small (up to 18% in [55]) improvement in transmission delay.

While HoL blocking can be avoided in TCP by using multiple TCP connections for each 'stream', there is increased overhead, as a TCB must be maintained for each connection (compared to one for the single multistreamed SCTP association). If the server has insufficient memory, incoming TCP connection requests may be dropped. This can also happen if the server runs out of ports. In addition, an application utilising multiple TCP connections could get an unfair share of the network bandwidth between the two hosts [115].

Establishing a separate connection for each 'stream' also increases the connection set up delay, as each TCP connection must perform a 3WHS. In SCTP, up to 65,535 independent streams can be established between two endpoints with a single four-way handshake association initiation.

3.4.1 Association Initiation

As SCTP is connection oriented, an association must be established before data can be exchanged. SCTP's association establishment is similar to TCP's SYN-cookie mechanism; SCTP uses a four-way handshake to protect servers from SYN flood DoS attacks. Figure 3.12 illustrates the association establishment process [114]. The client initiates an association by sending an INIT chunk. This chunk contains the required number of inbound and outbound streams (as streams are unidirectional), an initiation tag, the initial Transmission Sequence Number (TSN) and a list of all the IP address which the client can be accessed from. The initiation tag is a randomised 32-bit integer and is used as the verification tag during data transfer. This protects against blind 'man-in-the-middle' and sequence number attacks. It is also used to prevent old duplicate packets from previous associations from being processed as part of the current association.

The server responds with an INIT-ACK chunk. At this point TCP would reply with a SYN-ACK, and allocates resources; SCTP does not allocate any state. In addition to containing the parameters in an INIT chunk, a digitally signed cookie is also included. The cookie contains all the information required to establish an association, such as data in the received INIT chunk as well as the outgoing INIT-ACK, timestamps and TTL values. These are combined with a MAC or digital signature. As it is the server which ultimately processes the cookie, potentially anything can be included in the cookie, as long as the cookie can be successfully authenticated.

The client responds with a COOKIE-ECHO chunk, sending the cookie back to the server. The server validates the echoed cookie to check its authenticity. If the cookie is valid and has not been tampered with, the server considers the client and its association request legitimate. The cookie contains all the

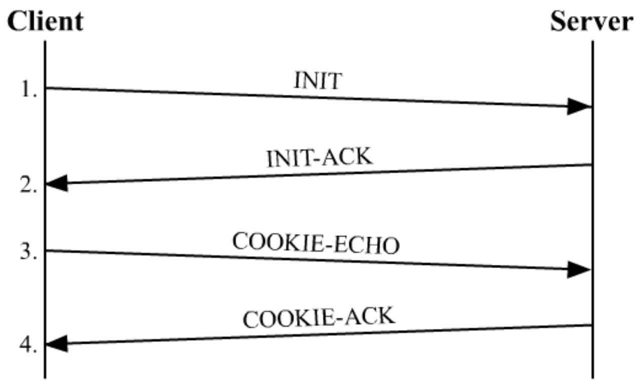


Figure 3.12: SCTP's four-way association initiation handshake. Use of a cookie protects servers from DoS attacks.

information required for an association, and only when the server receives and authenticates it, are resources allocated for the TCB. When the resources for the association have been allocated, the association is considered established on the server's side. This enables the segment containing the COOKIE-ECHO chunk to also contain data chunks as well (assuming the segment is sufficiently less than the MTU). This results in a connection establishment delay of one RTT, the same as TCP.

The server responds with a COOKIE-ACK chunk, in order to let the client know the cookie was received and valid. When the client receives this, the association is declared established on its end.

3.4.2 Association Shutdown

Unlike TCP, SCTP does not support 'half-closed' connections, where one endpoint shuts down while the other continues sending. Because of this reduced complexity, SCTP's shutdown is a three-way sequence, compared to TCP's four-way shutdown handshake. When an application wishes to close an association (Figure 3.13), no more application data is accepted by the transport layer and all queued data is transmitted. After this data is acknowledged, the client sends a SHUTDOWN chunk. When the server receives this, the same is performed: the application is notified, the transport layer stops accepting new data and any queued data is transmitted. When this remaining data gets acknowledged, the server sends a SHUTDOWN-ACK chunk. When the client receives this, the association is closed on its end, and responds with a SHUTDOWN-COMPLETE, closing the association on the server's side. If any of these shutdown chunks are lost, they are simply retransmitted.

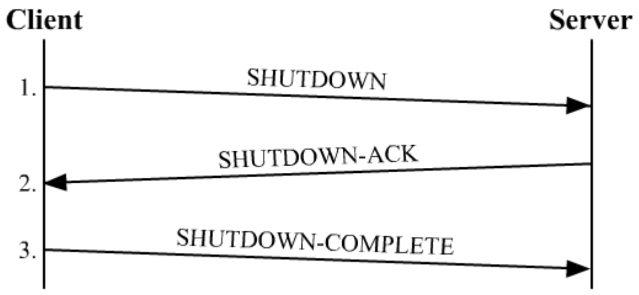


Figure 3.13: SCTP's shutdown sequence. Unlike TCP, closing an SCTP association closes it on both ends.

3.4.3 Implementations and Development

SCTP has kernel implementations[5] on Linux, Free/Open BSD, NetBSD, Solaris as well as many user space implementations. Many products currently use SCTP, with most of these being commercial SS7 signalling platforms; Cisco, Hewlett-Packard among others [53] have SCTP-based products to transport SS7 traffic over IP. Several end user applications[6] exist which have implemented SCTP, such as Mozilla and Apache. Unfortunately the Mozilla available for download supports *only* SCTP, therefore its usefulness is extremely limited as SCTP is not widely supported in webservers.

SCTP features in many research topics; current areas of research are described in [6, 71, 53, 113].

Like TCP, SCTP can be secured with IPsec [25] and TLS [72]. RFC 3554 describes IPsec support for SCTP's multihoming addresses. RFC 3436 describes TLS over SCTP, enabling applications over TLS to use multistreaming and multihoming. Other efforts to secure SCTP exist, such as Secure SCTP (S-SCTP)[64] and Datagram Transport Layer Security[7] for SCTP[63].

S-SCTP exists to overcome the scalability problems when TLS or IPsec is used. With TLS, each stream requires its own tunnel. Use of TLS also prohibits the use of several SCTP options, such as unordered delivery and partial reliability. With IPsec, a security association (SA) must be created for each IP address for multihoming. These will be discussed in Section 4.

[5]`http://www.sctp.org/implementations.html`

[6]`http://www.sctp.org/download.html`

[7]Datagram Transport Layer Security (DTLS) is a variant of TLS for use with datagrams and is defined in RFC 4347.

SCTP itself is still being optimised, with updates to its checksum algorithm [116], as well as an Implementers Guide [112]. SCTP options and extensions are also being developed:

Partial Reliability Extension ([42]): This enables SCTP to provide an unreliable, unordered service akin to UDP, as well as an ordered, unreliable service.

Dynamic Address Reconfiguration ([92]): This allows adding network interface cards without restarting the SCTP association.

Authenticated Chunks ([77]): Each SCTP packet contains a 32-bit verification tag (set to the same value as the initiation tag during association establishment) to help protect against blind 'man-in-the-middle' and sequence number attacks. It is also used to verify whether a packet belongs to the current association or a previous one. Authenticated chunks allow the sender to sign chunks, so the receiver can verify the authenticity of the sender and the data. It goes one step further than TLS, as TLS only secures application data.

PAD Chunk ([78]): This enables SCTP to discover the path MTU.

While SCTP has many attractive features and can be used in place of TCP (for example for a web browser), it is not a replacement for TCP. Previous work found that SCTP had lower overall raw throughput and higher latency than TCP [93]. SCTP's minimum retransmission timeout, a recommended value of 1000ms, was deemed too high and was considered a contributing factor to the slower performance. Another reason was that SCTP's message orientation (via chunks) added overhead. This overhead, however, becomes proportionately smaller when message size is increased, thus reducing the negative impact on SCTP performance.

When SCTP used only one stream, the performance was lower than TCP. However, when multiple streams were used, SCTP was able to outperform TCP.

As packet loss and network latency increased, the throughput gap between TCP and SCTP narrowed, and in some circumstances, the throughput and resulting latency equalled or exceeded that of TCP. Despite having lower performance (higher latency and lower throughput) than TCP, SCTP realises its original purpose, of providing a reliable transport for SS7 traffic.

3.4.4 Summary

SCTP's original purpose was to transport SS7 signalling messages over IP. Such a transport needed to meet the rigid timing and reliability requirements of PSTN signalling. Both the main transport protocols, TCP and UDP, have limitations which make them unsuitable for telephony signalling. SCTP was developed to address these limitations, however SCTP's features mean it is not limited to this scope and it is considered a general-purpose transport protocol.

SCTP's congestion and flow control are derived from TCP and is message oriented like UDP. SCTP improves reliability and resilience by supporting multihoming and a heartbeat system to detect link failures. SCTP avoids HoL blocking via multistreaming and out of order delivery. Another improvement over TCP is SCTP's four-way handshake initiation; SCTP has protection from SYN-flood type DoS attacks integrated into the protocol, the same as TCP's SYN cookies. SCTP can also be secured with several security architectures, including IPsec and TLS.

Implementations exist on several operating systems and SCTP is deployed in commercial SS7 signalling architectures. It also features in many research projects and continues to be refined and developed.

Given the features it brings, and in part combining the functionality of TCP and UDP, SCTP is a suitable candidate transport layer protocol for a secure transaction network.

3.5 Quick Transaction Protocol (QTP)

Quick Transaction Protocol (QTP) is a protocol designed specifically for multiplexing a high volume of short duration transactions (such as EFTPOS) over IP. The requirements of an application such as EFTPOS are in Section 2.6, and QTP meets these particular requirements well (the QTP specification was in fact used in part to create these requirements). QTP, created by Alcatel, Ascend (now Lucent) and INETCO [3], is not an RFC and remains as an IETF draft [43]. QTP behaves closest as a session layer protocol, however does not fit neatly into that classification as it implements some transport layer functionality as well.

Because QTP is designed for low latency message transfer, UDP is the preferred transport protocol. QTP, however, can be run over TCP, as it is independent of the transport layer. The specification assumes there is a Network Access Server (NAS) between terminals and the transaction processor (Figure 3.14), however terminals can have a direct connection to the network server. Terminals usually initiate sessions to the transaction processors, however transaction processors can also initiate connections to terminals. Reasons for this include requesting audit information from terminals, or sending stolen "hot card" lists to terminals [43].

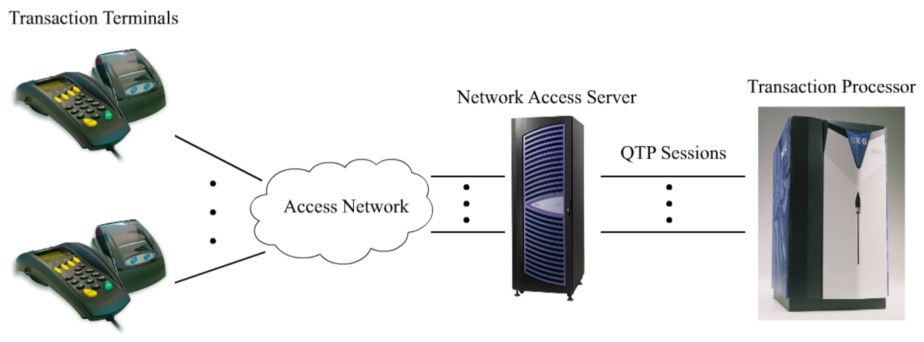


Figure 3.14: QTP assumes a Network Access Server is positioned between Transaction Terminals and Transaction Processors. QTP also permits Terminals to connect directly to the Transaction Processor.

Each logical connection is a QTP session operating over UDP, using a single port. Using UDP has the benefits of message orientation (as opposed to stream oriented as TCP is) and low overhead (UDP is a lightweight protocol and does not require a connection establishment or teardown phase as TCP does).

When an application uses UDP for its performance and simplicity, but requires reliability and congestion control, the application must implement its own transmission and flow control mechanisms, and QTP is an example of this. QTP achieves reliability over UDP (or other unreliable transports, as QTP is transport independent) with request-response message pairs.

QTP also has a rudimentary flow control system as part of its network status mechanism. Flow control in QTP refers to "the ability of a QTP entity to service further transactions with a remote QTP Entity" [43], unlike in TCP and SCTP, which refers to changing the sending rate when network congestion causes packet loss. A QTP entity has four states:

Available: operational and able to accept new QTP sessions.

Partially Congested: is nearing capacity, but can still accept new QTP sessions.

Congested: at capacity and unable to accept new QTP sessions. Existing sessions can continue without interruption however.

Shutdown: the QTP entity is unavailable and cannot accept new QTP sessions. Any existing sessions must be terminated immediately.

These flow control states are carried in Status Request and Report messages (Figure 3.15). Status Report messages can be sent periodically as a heartbeat/keep alive mechanism, as well as in response to a Status Request. If an entity fails to receive a Status Report when one is requested, it can either resend indefinitely, or declare the session down. When this happens, the entity terminates all sessions with the entity and attempts to restart. Status messages can also contain an advertisement of itself as a secondary QTP entity (with an attribute called Station Status, with a value of 1 for Primary or 2 for Secondary). Secondary QTP entities are used when the Primary QTP entity fails, for example due to overloading. This provides multihoming like behaviour. Unlike SCTP, which can have multiple alternate addresses and uses mutlihoming for redundancy and network resilience, QTP has the entities advertise themselves as Primary or Secondary entities, and uses this feature for load balancing. Ping (to determine entity existence and network timing) and Call State (to determine the state of a QTP session) attributes can also be carried in Status messages.

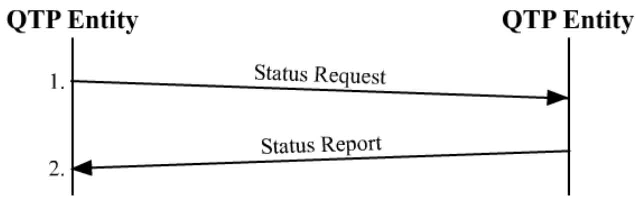


Figure 3.15: QTP Status Request and Report messages are used to report the operational capacities of QTP entities.

3.5.1 QTP Startup

Before a QTP entity can create or receive QTP sessions, and when using UDP or an unreliable transport, the draft recommends it request the status of the remote QTP entity (Figure 3.16). This is called Safe Startup. The QTP entity periodically sends Status Requests with a Flow Control value of “Shutdown”. When a Status Report is received, the entity then periodically issues Status Requests with a Flow Control value of “Available”. When this is acknowledged with a second Status Report, start up is complete, and the entity can start creating or receiving QTP sessions. When a reliable transport is used, QTP can use a mode called Quick Startup, in which QTP sessions can be created and received immediately, that is, it is not required to request the status of the remote entity; it is only recommended.

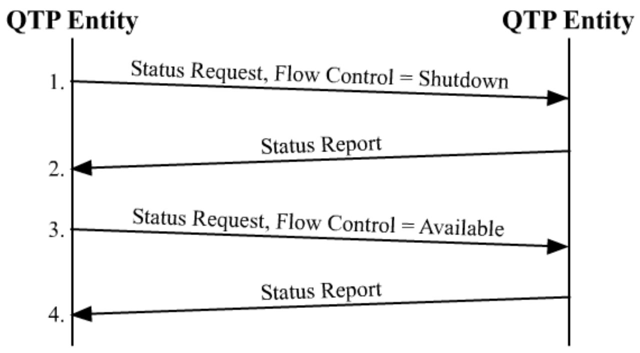

QTP Entity
QTP Entity
1.
Status Request, Flow Control = Shutdown
Status Report
2.
3.
Status Request, Flow Control = Available
Status Report
4.

Figure 3.16: QTP "Safe" Startup sequence. This is used when the transport protocol is unreliable.

3.5.2 Session Startup (Call Request)

Session establishment in QTP is a simple Call Request and Call Ack or Call Reject pair (Figure 3.17). A Call Reject contains, as an attribute, the reason for rejecting the call. If there is no response from the remote entity, the request is sent again. If the second attempt fails, the application is notified.

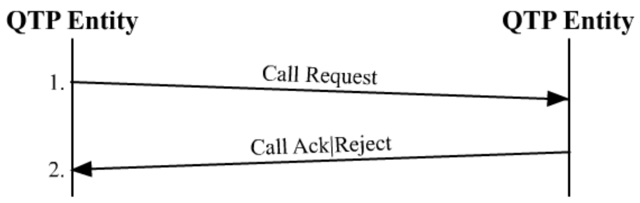


Figure 3.17: QTP Call establishment.

3.5.3 Data Transfer

Data can be sent with or without requiring acknowledgements. Acknowledgements, in addition to indicating successful delivery, can be withheld to throttle the sender. Figure 3.18 shows a data transfer that requires acknowledgement. An acknowledgement is simply sent in response to data. If no acknowledgement is received (when one is required) the original message is retransmitted. If the retransmitted message is also unacknowledged, the call is cleared (the session is torn down) and all session resources are released. Data that does not require acknowledgement is simply Figure 3.18 without the acknowledgement.

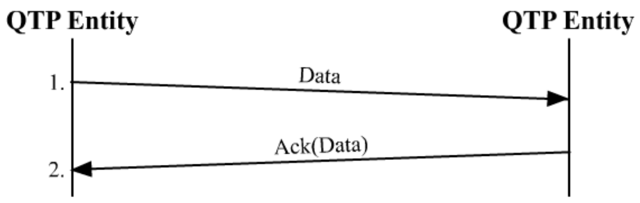


Figure 3.18: QTP Data transfer with acknowledgements. Acknowledgements are optional.

3.5.4 Session Shutdown (Call Clear)

To terminate the session (Figure 3.19), the entity sends a Clear Request (along with a reason as an attribute). This Clear Request may also contain transaction data. A Clear Ack is sent in response, clearing the session. If no acknowledgement is received (or the Clear Request was lost), the entity resends the Clear Request, and all session resources are released.

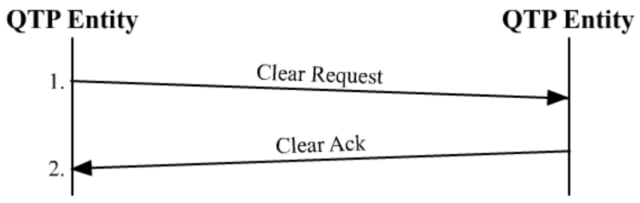


Figure 3.19: QTP Call Clearing.

3.5.5 Implementation and Development

The only available public implementation[8] is obtainable only by request, via email to qtp@inetco.com. The implementation is for Windows, and dates back to 1998. The last draft expired in May 2005, therefore the protocol cannot be considered still under public development.

[8]QTP is available under the open source model as stated in `http://www.inetco.com/technology/qtpfaq.html`

Several commercial products implement QTP, although likely to be in a more advanced state than the publicly available implementation from 1998. INETCO[9] System's BankLink, POSway and INETCO Connect protocol converter products support QTP. Vcomm's Vista[10] line of network access devices use QTP to transport EFTPOS over IP based networks (such as GPRS and Ethernet). Lucent also supports QTP in its MAX TNT dial gateway[3]. These companies were also contributors to the QTP specification.

As for deployment in current systems, it is used in Telstra's Argent system which replaced the Transend network.[11] PetroCanada also uses QTP based systems to transport EFTPOS to backend transaction switches.

Other than the internet draft specification, there is not much public information available regarding QTP. This gives the QTP protocol leverage to be used as a competitive advantage by the companies that conceived it. This and the specialised application for which QTP was developed has meant the companies involved in its development have not pursued standardisation. This instils a somewhat proprietary nature upon QTP, despite the availability of a stagnant implementation. As stated in Section 2.6, proprietary systems have several disadvantages. An organisation, which is not partnered with one of QTP's developers has no guarantee that its own implementation can interoperate with existing systems as the protocol was never standardised; extensions may have been specified but kept private within the developing companies.

3.5.6 Summary

QTP was developed specifically for transporting transactions (such as EFTPOS) over IP. Consequently it features low latency, reliable message transfer, with flow control and load balancing mechanisms. QTP is designed to run over UDP in order to utilise the protocol's performance benefits. Reliability is built into QTP itself, however QTP can also run over reliable protocols such as TCP.

A public implementation exists for Windows, however it has not been updated since 1998. Commercial implementations exist, however the products are produced by companies involved in QTP's specificiation. Given QTP's somewhat proprietary nature, it is not aligned with the goals of building an open, standards based architecture. Consequently, it cannot be considered a suitable component for this project.

[9] `http://www.inetco.com`

[10] `http://www.vcomms.com/vista.html`

[11] Incidentally QTP was originally designed for the Argent network.

3.6 Session Initiation Protocol (SIP) - RFC 3261

Session Initiation Protocol (SIP) is an application layer protocol used to establish, modify and terminate multimedia sessions. SIP can operate over TCP and UDP as well as SCTP [101]. As SIP provides its own reliability and retransmission mechanism, it can operate over unreliable transports such as UDP. Like QTP, it is an example of implementing reliability at the application layer. Note, however, that no congestion control is implemented in SIP; SIP is a session layer protocol, not a transport protocol.

SIP is defined in RFC 3261 [102] and is also extensively described in [32]. Accordingly its operation will only be described briefly.

There are a number of network entities defined by RFC 3261.

User Agents (UA): the two 'final' endpoints of a SIP session. Users interact with each other via user agents, and these may reside in a VoIP or conference application running on a computer or a hardware device such as a SIP phone.

Redirect Server: in response to invitations, a redirect server responds with a list of SIP Uniform Resource Identifiers (URIs[12]) or the address of another server which may know where the user is located (Figure 3.20). Here, two invitation requests are sent from the caller; one to the domain server, Company.com (a redirect server, which responds with the location of the callee) and one to computer2.company.com.

Proxy Server: a proxy routes requests on behalf of the UA, either to another proxy server, or the destination UA. For example in Figure 3.21, the domain server (Company.com) is a proxy server. If the user is located at computer2.company.com, but the invitation request was sent to Company.com, the invitation would be proxied to computer2. With a proxy server, the caller only needs to send an invitation to one location. Proxy servers can also provide mid-call features and enforce policy (for example limiting the destinations to which calls can be made)

Registrars: accepts REGISTER messages sent periodically from UAs. REGISTER messages notify the registrar of where the user is currently logged in, for example in Figure 3.22, caller@company.com can be contacted at

[12] A SIP URI is used to identify a SIP user, such as `sip:caller@company.com`. Company.com is the domain of caller's SIP service provider. URI's are similar in form to email addresses. A secure URI, SIPS, also exists (for example sips:caller@company.com). A SIPS URI requires TLS be used to transmit SIP messages.

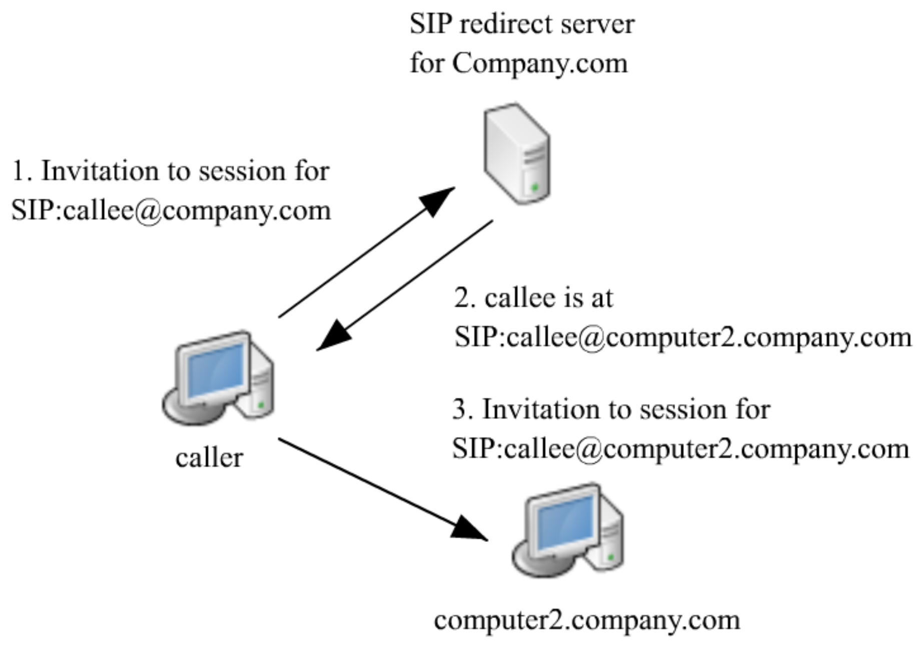

SIP redirect server
for Company.com
1. Invitation to session for
SIP:callee@company.com
2. callee is at
SIP:callee@computer2.company.com
caller
3. Invitation to session for
SIP:callee@computer2.company.com
computer2.company.com

Figure 3.20: A redirect server returns a list of possible locations where the user may be. The caller then sends another invitation to where the user is located.

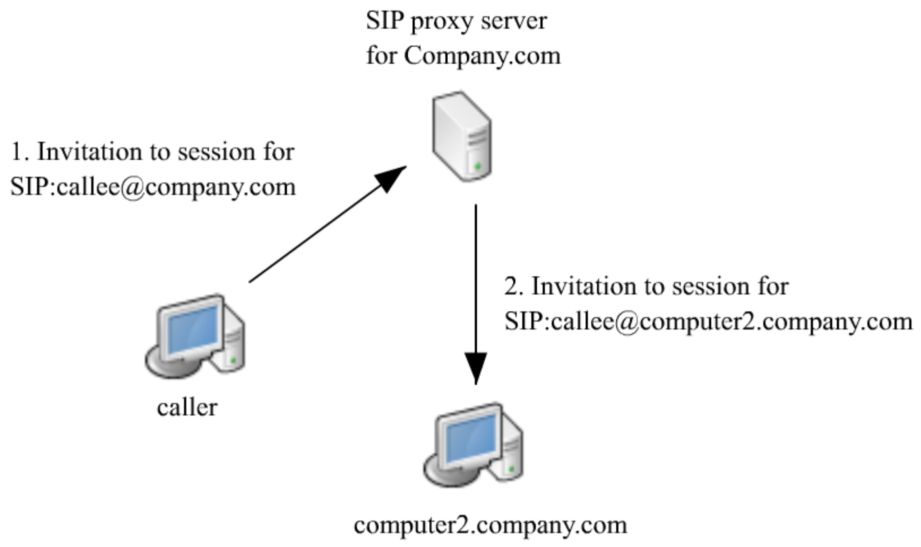

SIP proxy server
for Company.com
1. Invitation to session for
SIP:callee@company.com
2. Invitation to session for
SIP:callee@computer2.company.com
caller
computer2.company.com

Figure 3.21: A proxy server forwards requests on behalf of the caller.

caller@computer1.company.com. This binding is stored in the domain's location server. A user can register from multiple locations, for example callee's voicemail server could also register, so when callee is busy with a call, incoming call requests can be directed to voicemail.

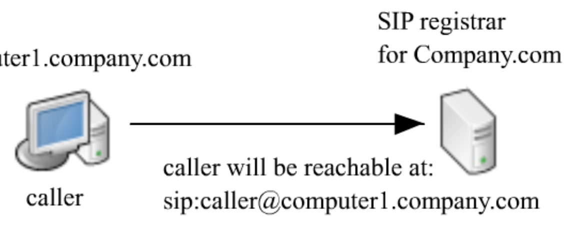

SIP registrar
for Company.com
computer1.company.com
caller will be reachable at:
sip:caller@computer1.company.com
caller

Figure 3.22: caller registers itself at computer1 with the SIP registrar.

Location Server: While not a SIP entity (as it does not use SIP to communicate with other SIP servers), location servers are used by a redirect or a proxy server to determine the location of a user. Registrars use the location server to store a user's location. Users can register from multiple locations, and the location server returns each location when queried.

These servers may be located in one physical server, that is, one server can potentially play all four roles.

3.6.1 Session Initiation

SIP can operate over UDP, TCP and SCTP, however UDP is most widely used [33]. Figure 3.23 illustrates the session initiation sequence. The caller sends an INVITE request to the callee's SIP URI (callee@companyB.com). This INVITE request contains information such as the caller, the callee, as well as information about the type of session. SIP does not actually provide session description attributes (such as type of media, codec, and bitrate); these attributes are described using another protocol such as Session Description Protocol (SDP) for a multimedia conference; a network game may use its own protocol for example. SIP separates the initiation of a session from its description, thus any format of session description can be carried in the SIP payload.[13]

As the caller does not know the location of the callee, or the callee's domain's SIP server, the INVITE is sent to the caller's SIP server, companyA.com (a proxy server). CompanyA.com responds with a `100 Trying` to the caller. This

[13]RFC 3261 describes the SIP payload in a SIP message as comparable to an attachment in an email message.

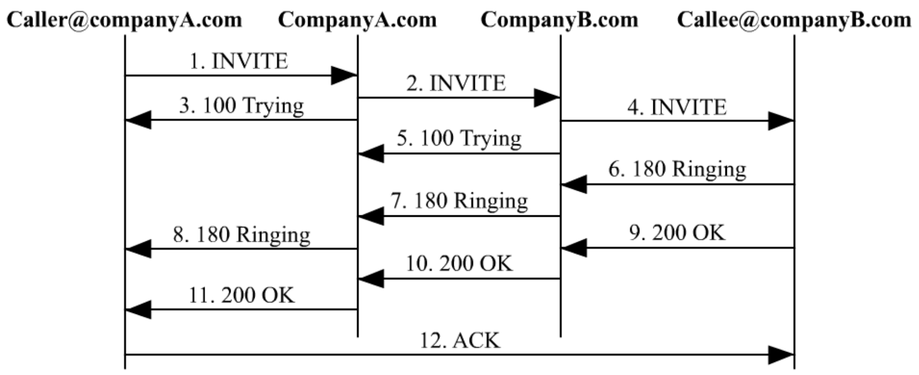


Figure 3.23: SIP session initiation sequence.

notifies the caller that the INVITE request was received, and is being forwarded on the caller's behalf to the destination. If no `100 Trying` is received and an unreliable transport is used, the INVITE would be retransmitted (at the application layer).

CompanyA.com locates the callee's SIP server at CompanyB.com (for example via DNS). CompanyA.com adds itself as a `via` attribute in the INVITE, and proxies the INVITE request to CompanyB.com. The `via` attribute indicates that responses should be sent back to this proxy. When the CompanyB.com proxy receives this, it also responds with a `100 Trying` back to CompanyA.com (this second `100 Trying` is not passed back to the caller as it is what RFC 3261 terms a 'provisional response'). CompanyB.com uses the generically named location service (possibly provided by a location server) to locate the callee. CompanyB.com adds itself as another `via` attribute, and passes the INVITE to the callee.

The proxy server could also fork the INVITE request. If the callee had registered itself from more than one location, CompanyB.com could send the INVITE request to each possible location; this is known as forking. This provides a degree of multihoming, as a single UA can register from multiple locations (these locations can be logical, for example the secondary network interface card on a host).

The callee's UA alerts the user, for example a SIP phone would ring, or a conferencing application could pop up with a message indicating a user wishes to start a conference session. The UA responds with a `180 Ringing` message, which is passed back to the caller through the proxies by sending back to the first address in the `via` attribute (CompanyB.com removes its `via` attribute and passes it to the next `via` attribute and so on). Being a 'provisional response', the `180 Ringing` message is not reliably transmitted back to the caller. If the caller does receive it, however, it can notify the user.

If the callee answers the call, the UA sends a `200 OK` through the proxies back to the caller. A `200 OK` is termed a 'final response' and is sent reliably. This message contains a description of the session the callee can support, carried in the same way as in an INVITE message. The `200 OK` message also contains a URI header, of the callee's exact location.

When the caller receives this, the UA notifies the user (for example stopping the ringing tone) and sends an ACK. This ACK is sent directly to the callee, bypassing the two proxies. Both the caller and callee have learned each other's address; they are carried in the INVITE and `200 OK` messages respectively. At this point, the proxies do not need to be in the message exchange path, as each user knows the other's location. However, proxies can choose to remain in the exchange path to provide mid-call features.

This is called a three-way handshake, as only three messages (INVITE, `200 OK` and `ACK` are sent reliably. When this handshake is completed, the session described in the SIP body (for example using SDP) can begin. RFC 3261 states the media packets can take a different path from the SIP signalling messages. This is similar to SS7 in a PSTN, where the signalling and voice take a different path [32].

3.6.2 Session Modification

Characteristics of the media can be modified mid-session (Figure 3.24). An INVITE sent during a session is known as a re-INVITE, and references an existing session so users do not have to establish a new session. If the change is not accepted, a `488 Not Acceptable Here` is sent and the session continues with its existing attributes, otherwise a `200 OK` accepts the change.

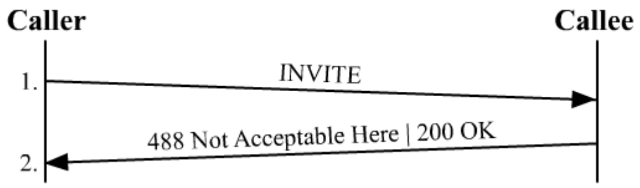


Figure 3.24: SIP session modification.

3.6.3 Session Termination

Sessions are terminated with the BYE message. In Figure 3.25, the callee ends the session, and the caller responds with a `200 ACK` message that terminates the session. Strictly speaking, a BYE message indicates a user has left the session. In a two user scenario, the session would be terminated, however in a multicast conference, the session would not be affected as long as users were still in the conference.

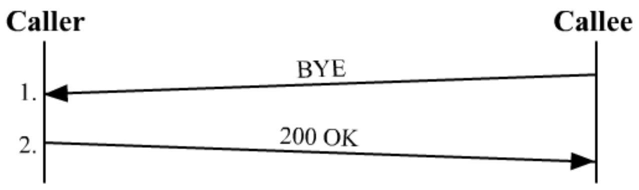


Figure 3.25: SIP session termination.

3.6.4 Implementation and Development

SIP is used in many applications, from instant messaging applications such as PhoneGAIM to VoIP networks such as Gizmo. Cisco and Nortel among others have SIP server platforms.[14] SIP-based Private Branch Exchange (PBX) telephone networks are available from Asterisk and 3Com among other vendors.[15] As the SIP standard is well established, most activity is focussed on creating products that are interoperable. SIP also features in many research projects, ranging from mobile VPN and running VoIP through firewalls and NATs to applying SIP in new ways.

The H.323 standard, regulated by the International Telecommunication Union (ITU) provides a similar function to SIP; H.323 is a signalling protocol for IP networks. H.323 is older than SIP, however, and has a 'monolithic' architecture [5], for example H.323 specifies everything from codec and session description to QoS. This makes it difficult to add new or modify existing components. SIP is considered simpler and more flexible as its core responsibilities are providing call management (set up and termination) and user location. All other functions such as QoS and session description are provided by other protocols. SIP's design has meant it can operate with HTTP and SMTP [4]. This enables SIP to provide internet services such as sending an email to unreachable users.

[14]`http://www.SIPcenter.com/SIP.nsf/html/SIP+Servers`
[15]`http://www.SIPcenter.com/SIP.nsf/html/IP+PBX`

SIMPLE

The Instant Messaging and Presence Protocol (IMPP) Working Group existed[16] to define a generic Instant Messaging and Presence (IM&P) model such that independently developed instant messaging (IM) applications could interoperate with each other. A document has been produced [44], which specifies the minimum requirements such a protocol must meet.

The SIP specification (RFC 3261, [102]) contains a rudimentary presence feature in the form of periodic REGISTER messages. Using this feature for presence, however, is rather crude as it is not its intended purpose.

The SIP for Instant Messaging and Presence Leveraging Extensions Working Group (SIMPLE) defines a set of extensions that provides SIP with instant messaging and a presence service. This is particularly applicable to this project; instant messages are transferred in real time and the presence service provides a degree of 'network intelligence' (where the presence and availability of a remote device or peer can be determined). Having knowledge of the network conditions can improve reliability.

SIMPLE, alongside the XML based Extensible Messaging and Presence Protocol (XMPP) are the two proposed IMPP's developed by the IETF [36]. As [36] explains SIMPLE, it will only be briefly described here.

Presence

Presence in SIMPLE [99] defines a Presence User Agent (PUA), or a presentity [45] which provides presence information to a presence server (or service). UserB (a PUA) REGISTERs its current status with the presence server in Figure 3.26 (such as "Available") upon initialisation, as well as whenever the current status changes.

UserA wants to know UserB's current status, and SUBSCRIBES to UserB's presence information. Each time UserB's status changes, the presence server transmits a NOTIFY to each subscriber. When the subscribers receive this, they are updated on the user's current status. This behaviour closely resembles MQTT as described in Section 2.5, where the presence server is the message repository and the PUA is the remote device.

Instant Messaging

SIMPLE also defines a MESSAGE method in RFC 3428 [34], where a message can be contained in a SIP body. A MESSAGE request is treated as a BYE

[16]The group concluded in September 1998.

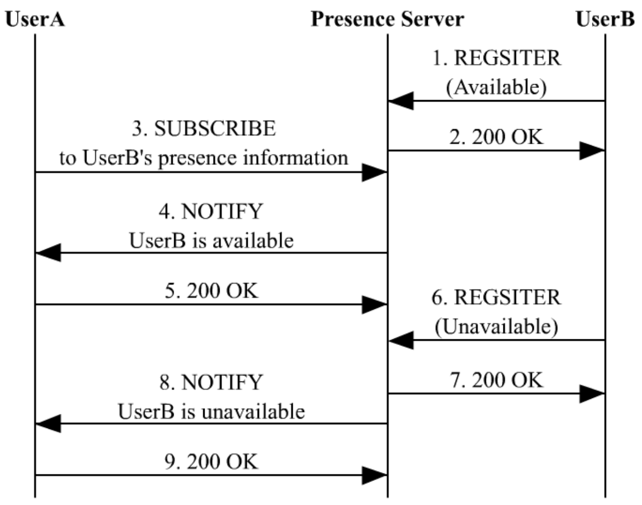

UserA
Presence Server
UserB
1. REGSITER (Available)
2. 200 OK
3. SUBSCRIBE to UserB's presence information
4. NOTIFY UserB is available
5. 200 OK
6. REGSITER (Unavailable)
7. 200 OK
8. NOTIFY UserB is unavailable
9. 200 OK

Figure 3.26: UserA registers to be notified of UserB's status. When UserB's status changes, all subscribers are notified with a NOTIFY message.

request by proxies [32]. The advantage here is existing SIP infrastructure can be reused to provide IM capabilities without any modification. Figure 3.27 illustrates this.

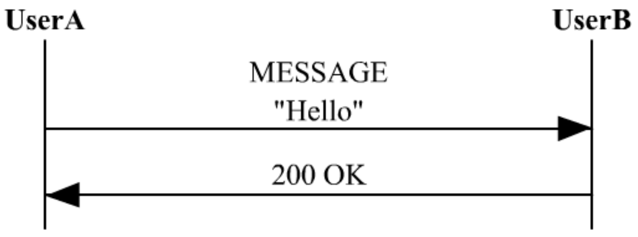


Figure 3.27: UserA sends an instant message (IM) to UserB.

These extensions to SIP would benefit a transport protocol that is not aware of its own state of service, such as TCP. While TCP contains a keep alive function (defined in RFC 1122 [29]), its implementation is optional, as it is not in the TCP specification (RFC 793 [97]).

Multihoming can also be improved with these extensions; instead of forking a request to every registered location for a user, a proxy server would only need to forward requests to where the presentity is declared available. This would give similar resulting functionality to SCTP's multihoming, only here it is at the application layer.

3.6.5 Summary

SIP is an application layer protocol used for session establishment, modification and termination. SIP is independent of the type of session created and these sessions can be any type, from voice to video. SIP is part of a modular architecture, and can therefore be used in conjunction with other services to provide new functions. SIP has reliability built in and is independent of the transport protocol, so can operate over unreliable transport protocols.

SIP is currently used in many applications and features in many research projects, leading to new ways to apply SIP. The SIMPLE extensions are relevant to this project, as they provide an instant messaging functionality as well as a degree of network service awareness through the presence facility. The benefit of providing these services at the application layer is that any transport protocol can be used to carry SIP.

A lot of activity surrounds SIP and many products will be deployed based on SIP. SIP is an open protocol, and because of its simplicity, can be easily extended and applied in many new ways. This flexibility and in particular the SIMPLE extensions make SIP a worthwhile component for this project.

3.7 Simple Object Access Protocol - SOAP

Simple Object Access Protocol (SOAP) is a protocol based on XML and HTTP that enables applications to invoke procedures on remote hosts and exchange information. SOAP is a W3C standard and is up to version 1.2.[17] It is part of the web services architecture [20], comprising SOAP, Web Service Description Language (WSDL) and the Universal Description, Discovery and Integration (UDDI) registry.

WSDL uses XML to describe the capabilities and locations of web services, such as the message format and parameters for the function or method [8]. The interfaces and services described by WSDL may be registered in a UDDI directory [37]. Businesses can also search for web services in the UDDI directory service. This enables business partners to share information more efficiently as interoperability and integration of services is easier; W3CSchools[18] uses a flight reservation example. Airlines register their flight rate and reservation checking systems into a UDDI directory. Travel Agents then search the directory for an airline's interface description. The travel agency can then immediately use the airline's interface.

[17] `http://www.w3.org/2000/xp/Group`

[18] `http://www.w3schools.com/wsdl/wsdl_uddi.asp`

Web services and e-commerce sites are usually built using a three-tier architecture: a front end web server, an application server in the middle and a backend database. As SOAP uses HTTP[19], SOAP requests are sent to the organisation's web server. These requests are routed to the application server, where the SOAP processor resides, and parsed.

Frameworks exist so normal applications can be translated into a SOAP based web application (for example Microsoft's .NET and Sun's Enterprise Java Beans). This makes SOAP requests language independent, as the SOAP request is translated and parsed by the SOAP/application server into the native language. Responses are sent the same way as a SOAP response through the web server.

As SOAP is based on XML, it is platform independent. The application frameworks make it language independent as well. These contribute to SOAP's interoperability. SOAP uses web servers to communicate, and this can be considered an advantage. Businesses have experience deploying web servers and web applications [80] and can utilise existing infrastructure. Firewalls permit traffic on port 80 so the web server can receive requests. As SOAP uses existing web servers to communicate, firewalls do not need to open extra ports to accommodate SOAP; SOAP uses the web server's opened ports. With other remote access architectures such as Remote Procedure Call (RPC), Common Object Request Broker Architecture (CORBA) and Java's Remote Method Invocation (RMI), additional ports on the firewall need to be opened, and this increases risk. SOAP can operate unimpeded as the firewall sees SOAP requests as regular web page requests.

However, this behaviour ultimately circumvents the purpose of a firewall. Since SOAP requests appear as standard web traffic, the firewall now has no way of determining whether the traffic is a harmless web request, or an application request message that invokes a function on the server. Firewalls must then filter at the application layer by inspecting the packet contents. This commonly happens with email, where potentially harmful attachments are removed from messages as noted in [20].

While SOAP is a valid architecture for this project, its focus is on interoperability. This project's goal is to define a protocol stack, which would be employed universally throughout the entire transaction network. In a controlled deployment scenario, such as an EFTPOS system, all equipment and infrastructure would be authorised and sanctioned by the network operator. Consequently, interoperability problems would be avoided as the entire network would use the

[19]SOAP can also use SMTP

same application layer protocol, in this instance AS2805, as well as the same underlying network stack (the focus of this project). SOAP's emphasis on interoperability between different application vendors would be unnecessary in an environment such as this.

It is conceivable that a payment system such as EFTPOS could be composed of network infrastructure from different vendors, each using their own payment card standard. In a relatively uncontrolled situation like this, where the interface used by each vendor is unknown, SOAP would have value. Clients could determine the interface of the vendor via a UDDI lookup and integrate seamlessly. However, allowing unknown devices to integrate into a payment network and become part of the infrastructure would create security risks.

3.8 Versatile Message Transaction Protocol (VMTP) - RFC 1045

Versatile Message Transaction Protocol (VMTP) is an experimental transport protocol, defined in RFC 1045 [38]. VMTP is designed to support RPC and transaction oriented traffic. Each request and response has a transaction identifier. When a server receives a request, it locates the corresponding transaction record for the client. If no record exists, the server queries the client to obtain the necessary information. Thus VMTP establishes connections on demand [39], and no separate connection establishment phase exists, such as TCP's 3-way handshake.

Each network entity has a unique and stable 64-bit identifier that is independent of the IP address. This facilitates process migration, mobility and multihoming, as entities are no longer tied to an IP address.

Multicast, datagrams and security are other features of VMTP. Datagrams in VMTP are simply requests that do not require a response. Acknowledgements in VMTP are implicit [68]; a response implicitly acknowledges the request, and a subsequent request implicitly acknowledges the response. VMTP also offers a streaming mode resulting in similar functionality to TCP. A stream of requests can be issued with the responses received asynchronously. These responses are akin to a TCP ACK.

Regardless of these qualities, VMTP cannot be considered a suitable transport protocol; it remains in experimental state and RFC 1045 has not been updated since February 1988, nor has it been superseded by a newer document. VMTP perhaps reflects the best practices of the time in terms of protocol design; it is a function rich protocol, which the RFC states "provides high data rates, low error rates and relatively low delay". VMTP can even operate without IP; it can be layered directly on top of a data link layer protocol. It is a multipurpose protocol, in contrast with modern protocols. Current protocols are typically single purpose and lightweight, for example SIP and TLS.

3.9 Xpress Transfer Protocol (XTP)

Xpress Transfer Protocol (XTP) combines the functionality of TCP, UDP and TP4 [121]. While not an IETF RFC, XTP is defined in [10]. It has been adopted as part of SAFENET, a U.S. military standard for naval network communications [9]. XTP is not designed as a replacement for TCP, UDP or TP4, but to work alongside them. XTP, like VMTP, can operate over IP or directly over the data link layer (for example Ethernet or ATM). XTP is designed to provide low latency and high throughput for real time systems, for example weapons control, and generally has higher throughput than TCP [121].

XTP is centred around the separation of error, flow and data transmission control policies. These three properties can be configured independently of each other. XTP has three selectable error control modes: reliable (similar to TCP), unreliable (akin to UDP, where the sender does not receive acknowledgements) and fast negative acknowledgement. Fast negative acknowledgement means a receiver, upon identifying out of sequence packets, can immediately inform the sender of the missing data. The sender then retransmits only the missing data. This functionality is similar to TCP's SACK mechanism. Flow control is also selectable in XTP, and can either be traditional credit windows, a more conservative policy called reservation mode and disabled (used for streaming data, for example multimedia streams).

A receiver can control the data transmission of a sender by adjusting the rate and burst parameters. Rate control limits the amount of data that can be sent at a time while burst control determines the size of data that can be sent. Unlike conventional flow control which is end-to-end (with no router intervention), XTP's flow control can include the routers. This enables a loaded router to limit the rate of incoming traffic until the network is no longer congested.

XTP features multicast, priority/QoS and fast connection setup. XTP can achieve with three packets what TCP accomplishes with six. The first packet opens the connection, contains the data and also closes the connection. The second packet contains the response (if any) and the acknowledgement of the close request and the data. The sender to receiver connection is then closed. The second packet also contains a close request in the reverse direction. The sender acknowledges this with a third packet, which closes the receiver to sender connection. XTP also has a two packet exchange for transactions, where the sender sends a connection request and the data (the transaction request). The receiver responds with its data (the transaction response) and closes the connection

XTP was developed by the XTP forum, which has since disbanded. The latest revision of XTP is 4.0b [10] from 1998, and development on the protocol has stopped. A public implementation was once available from Sandia National Laboratories, as were commercial implementations from Network Xpress and Mentat, however these no longer exist; Network Xpress ceased operating in 1999 and Mentat was purchased by Packeteer in late 2004. Accordingly, XTP is an unsuitable protocol for this project.

Chapter 4

Security Candidates

This section introduces several security architectures which could support the security requirements defined in Section 2.6. The goal of any security protocol is to authenticate the end points as well as prevent eavesdropping through encryption and message tampering through data integrity protection.

It is important to note that no security protocol can stop all denial of service attacks or prohibit traffic analysis. These risks can be reduced, however, for example SYN cookies can reduce the impact of SYN flooding DoS attacks. Traffic analysis requires analytic skill to perform successfully. Knowing network traffic exists between two sites can be useful to an analyst, for example between a client and a transaction processor. Subsequent statistical analysis could be performed which may reveal patterns in the volume, direction and timing of traffic. This could suggest which end was a client and which was the transaction processor. However, resistance to traffic analysis can be improved by encrypting at the application layer, for example the application itself may encrypt the transaction data, before being encrypted at the transport layer. Aggregating several data streams onto a single stream can also make traffic analysis harder.

4.1 Secure Shell (SSH) - RFC 4251

The Secure Shell Protocol (SSH) is used for secure remote logins and tunnelling through an unsecure network. Various methods exist to administer remote systems from a console, such as Telnet, rsh and X [23]. However, these methods transmit in cleartext and do not provide authentication. SSH2 was designed in 1996 in response to security flaws in SSH1 [12], such as CRC checksum attacks. This made SSH1 vulnerable to Man-in-the-Middle (MITM) attacks. In 2006, the Secure Shell working group (Secsh) proposed SSH2 (which obsoletes SSH1) as an IETF proposed standard. The architecture is defined in RFC 4251 [125] and consists of three main components:

The Secure Shell (SSH) Authentication Protocol (RFC 4252)[123]: This describes the authentication protocol framework and describes the public key, password and host-based authentication methods. It runs over the SSH Transport layer protocol and authenticates the user to the server.

The Secure Shell (SSH) Transport Layer Protocol (RFC 4253)[126]: This runs over TCP/IP and thus is not a 'transport layer protocol' in the network stack sense. This protocol can provide a secure network service and offers encryption, data integrity and authentication as well as key exchange methods and algorithm negotiation. It provides only host-based authentication and relies on higher layers for user authentication.

The Secure Shell (SSH) Connection Protocol (RFC4254)[124]: This provides secure interactive login sessions, remote execution of commands and forwarding of TCP ports. These different logical channels can be multiplexed onto a single secure connection. This also operates over the SSH transport layer and authentication protocol.

SSH's port forwarding capability enables it to create secure TCP tunnels between endpoints. These tunnels can be created on any port, and can therefore be used to transmit data securely for any TCP application. Tunnels are created with the `-L` and `-R` options, used for local forwarding and remote forwarding respectively.

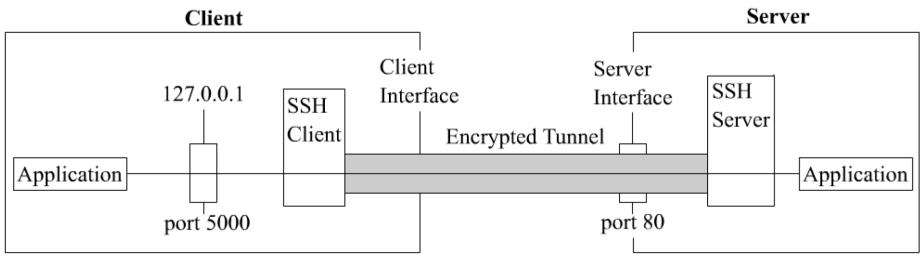


Figure 4.1: Anything sent to 127.0.0.1:5000 on the client is forwarded securely to server:80 by SSH.

Figure 4.1 shows a tunnel created between two hosts. Using local forwarding (the `-L` option), the tunnel would have been established on the client with the command:

```
client:$ ssh -L 5000:server:80 server
```

On the client, this command forwards anything directed to localhost (the client's 127.0.0.1):5000 to server:80 through the SSH secured tunnel.

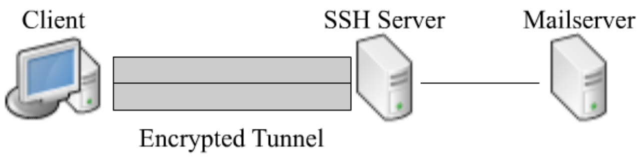


Figure 4.2: A tunnel created with the command `ssh -L 5000:mailserver:110 server`. The traffic is encrypted between client and server, but is in plaintext between server and the mailserver.

Connections can also be created to hosts behind the SSH server, for example in Figure 4.2. Using local forwarding again, the tunnel would have been established on the client with:

```
client:$ ssh -L 5000:mailserver:110 server
```

This command works as follows.

1. Upon typing the command on the client, the SSH client binds to port 5000 on the loopback address 127.0.0.1 on the client
2. When a process on the client connects to 127.0.0.1, the SSH client accepts the connection
3. The SSH client informs the server through the encrypted tunnel to create a connection to mailserver:110
4. The SSH client forwards any data directed to 127.0.0.1:5000 to the server through the encrypted tunnel.
5. The server decrypts the data and forwards the plaintext data to mailserver:110.
6. The mailserver sends responses back to the server in plaintext.
7. The server forwards these through the encrypted tunnel back to the SSH client.
8. The SSH client on the client decrypts these and sends the plaintext back to the process.

9. When the connection is closed, it is torn down in the tunnel as well.

With this method, only data between the client and server is encrypted; the data transferred between the mailserver and the server will be in plaintext. In a sense, the server is operating as a gateway, as all data leaving the server destined for the client is encrypted, and all data leaving the server destined for the mailserver is decrypted. A secure tunnel must be explicitly created between the server and every destination which requires encryption.

However, an encrypted tunnel can also be created between the SSH server and the mailserver. Note that while the data is encrypted between endpoints, the tunnel is not end to end between the client and the mailserver; the SSH server decrypts the data from the client and re-encrypts it before sending it to the mailserver and vice versa.

Remote forwarding (using the `-R` option) has the same effect, but the tunnel is initiated from the remote side. Where the `-L` option forwards data to a remote port, the `-R` option forwards data from a remote port to a port on the local machine.

4.1.1 Connection Setup

SSH runs over port 22, officially assigned by the Internet Assigned Numbers Authority (IANA[1]). Establishing an SSH connection consists of four phases: protocol version exchange, key exchange, user authentication and opening the channel. The first two phases are defined in RFC 4253 [126], while the user authentication phase is defined in RFC 4252 [123]. The final phase, opening a channel is defined in RFC 4254 [124].

The protocol version exchange phase consists of the two sides exchanging an identification string (Figure 4.3). This phase is used to establish whether both sides support SSH2 (indicated by `2.0` in Figure 4.3), as well as triggering compatibility extensions or notifying implementation capabilities (noted via the software version, `SSH_3.6.3q3` in Figure 4.3).

The key exchange phase negotiates the MAC and encryption algorithms as well as the key exchange method. The encryption key is also exchanged during this phase, which the standard states should have an effective length of at least 128 bits. RFC 4253 requires only the 3DES-CBC cipher with a 168-bit key, as all known implementations support it. The standard recommends AES128-CBC, as the 3DES 168-bit key has an effective length of 112 bits, less than the minimum. Other ciphers supported are the CBC modes of Blowfish,

[1]`http://www.iana.org`

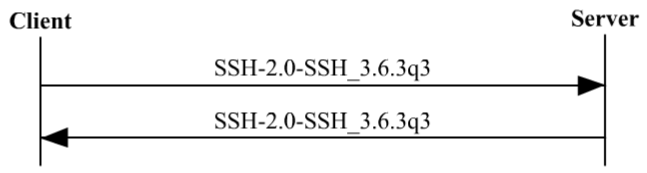


Figure 4.3: SSH Protocol Version Exchange. The client and the server exchange SSH version information; in this instance, SSH 2.0 is used, with SSH_3.6.3q3 the software version.

Twofish, Serpent, Arcfour, IDEA and CAST. As for MAC algorithms for data integrity, only SHA-1 is required, however MD-5 is also supported. Two key exchange methods are defined in RFC 4253: diffie-hellman-group1-sha1 and diffie-hellman-group14-sha1. These exchange methods define the generation of session keys used for encryption and authentication as well as the server authentication method. Additional methods may defined for each of the above three categories.

Figure 4.4 illustrates the key exchange phase. The SSH_MSG_KEXINIT message contains a list of all supported and preferred algorithms for each category (encryption, MAC and key exchange). Following this exchange, the selected key exchange method is carried out (in Figure 4.4, the Diffie-Hellman method is used, but other methods may be defined). The client sends a nonce value x (a function applied to a random number) in SSH_MSG_KEXDH_INIT.

When the server receives this, the server generates a nonce value y (also a function applied to a random number), and derives a session key (computed over the client's x and the server's y value). The server responds with its public host key, the value y (so the client can also derive the same key) and a digital signature in SSH_MSG_KEXDH_REPLY. The digital signature is hashed over the client and server's version strings, the two SSH_MSG_KEXDH_INIT messages, the server's public key, x, y and the derived key. This hash, H, is also used as the unique session identifier.

The client checks that the public key corresponds to the server (discussed in weaknesses, Section 4.1.4). The client also derives a session key computed over x and y then verifies the signature. If the server is authentic, the client and the server exchange the SSH_MSG_NEWKEYS message. This ends the key exchange phase and all subsequent messages use the newly negotiated keys and algorithms.

The key exchange phase establishes a session key, creats a secure channel and verifies the identity of the server to the client. The user authentication

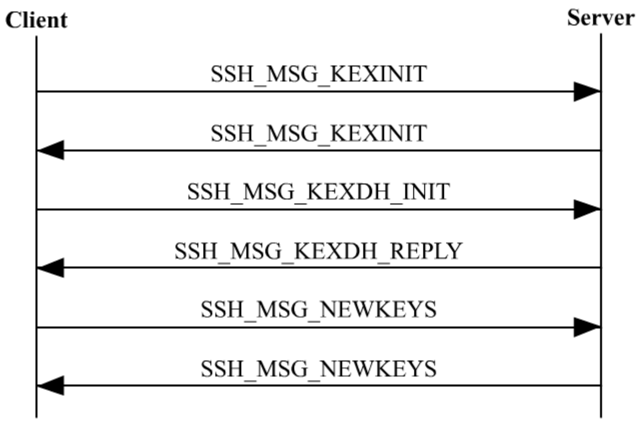


Figure 4.4: SSH Key Exchange. In this example, the Diffie-Hellman key exchange method is used, and uses two messages; messages 3 and 4. Other key exchange methods may use a different number of messages. When the SSH_MSG_NEWKEYS messages have been exchanged, all subsequent messages use the newly negotiated keys and algorithms.

phase verifies the user to the server. Three methods are available: Public key (required by RFC 4252), Password or Host-Based (both optional).

Authentication requests use SSH_MSG_USERAUTH_REQUEST messages and each of these messages contain the username on the client as well as authentication method specific fields. The server responds with SSH_MSG_USERAUTH _SUCCESS if the authentication is completed and a channel can then be opened. If the request failed, the server replies with SSH_MSG_USERAUTH_ FAILURE, with the partial success flag set to false. If the previous request was successful but further authentication is required, the server responds with SSH_MSG_USER AUTH_FAILURE with the partial success set to true.

The public key method authenticates based on the user's possession of a private key. Using the private key is relatively computationally expensive, so clients can check whether public key authentication is acceptable (Figure 4.5). This avoids unnecessary processing on the client. If it is acceptable, the server responds with an SSH_MSG_USERAUTH_PK_OK. The actual authentication (Figure 4.6) consists of the client sending the username, public key and a digital signature (a hash calculated over several values including the username, the user's public key, the session identifier[2] and encrypting it with the user's private key) to the server. The server checks whether the public key is a valid

[2]The hash H generated during the key exchange phase

authenticator for the user and the validity of the signature. If both succeed and no further authentication is required, the user is authenticated.

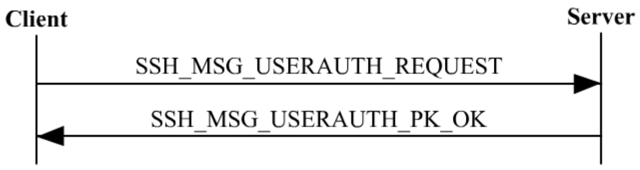


Figure 4.5: The client can check whether public key user authentication is acceptable, before performing computationally expensive public/private key operations.

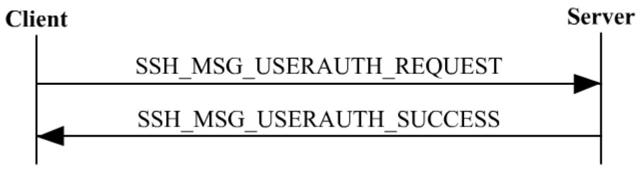


Figure 4.6: SSH User Authentication. While the actual content of the messages are different, the messages exchanged are the same for public key, password and host based user authentication.

The password method transmits the password in plaintext within the SSH packet, however all packets are encrypted since the successful key exchange. Figure 4.6 shows this method. If the password has expired, the server can request a password change (Figure 4.7) with the SSH_MSG_USERAUTH_PASSWD_CHANGEREQ message. The SSH client can use a different authentication method or request a new password from the user. The server may respond with either SSH_MSG_USERAUTH_SUCCESS if the password was changed successfully and the user is authenticated; SSH_MSG_USERAUTH_FAILURE with partial success if the password was changed but further authentication is required; SSH_MSG_USERAUTH_FAILURE without partial success if the password was not changed due to the server not supporting password changes or the old password was incorrect; or SSH_MSG_USERAUTH_CHANGEREQ if the new password is unacceptable (for example being too short).

The host-based authentication method authenticates on the username and the host the user is connecting from. This method is similar to the public key method, except it authenticates on the host's public key rather than the user's. The client authenticates by sending the username, hostname, the host's public

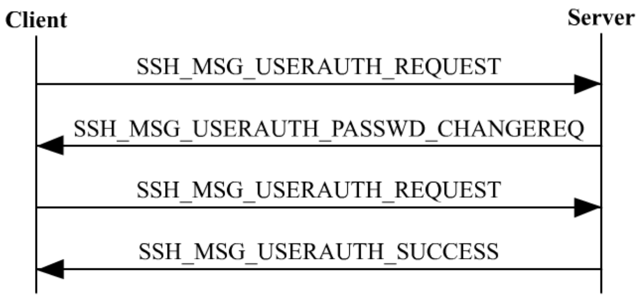


Figure 4.7: If an expired password is used to authenticate a user, the server can request the user to change the password.

key, the username logged onto the host and a digital signature calculated over these values and the session identifier (Figure 4.6). The server verifies the host's public key and validates the signature to complete authentication. The server may also check whether the user is authorised to log in.

Keys can be renewed at any time as long as a key exchange is not already in progress. Rekeying is identical to the initial key exchange with the exception of creating a new session identifier, which remains the same. The client or the server can initiate rekeying. RFC 4243 recommends rekeying after 1GB of data is transferred or one hour, which ever comes first. However, as public/private key encryption is used during the key exchange, it is computationally expensive, so should not be performed too frequently.

When the three phases of authentication are complete, a channel can be opened. Multiple channels are multiplexed on a single tunnel. Figure 4.8 illustrates opening a channel. A channel can be opened from either side, and in Figure 4.8, the client sends an SSH_MSG_CHANNEL_OPEN message, which contains a local channel number. The server replies with SSH_MSG_CHANNEL_OPEN_CONFIRMATION or SSH_MSG_CHANNEL_OPEN_FAILURE with a reason code of either:

- SSH_OPEN_ADMINISTRATIVELY_PROHIBITED
- SSH_OPEN_CONNECT_FAILED
- SSH_OPEN_UNKNOWN_CHANNEL_TYPE
- SSH_OPEN_RESOURCE_SHORTAGE

When a channel is opened, data transfer can begin using the SSH_MSG_CHANNEL_DATA message.

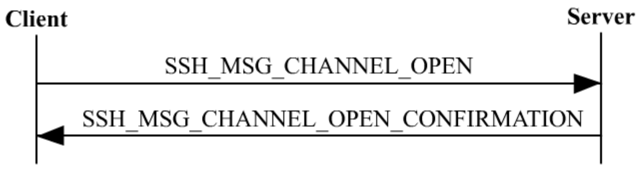


Figure 4.8: After the key exchange and user authentication phases, a channel can be opened, and data transfer can begin. Channels can be opened from either side.

4.1.2 Connection Closing

Like connection setup, closing a connection is performed at the connection layer (RFC 4254) and at the 'transport layer' (RFC 4253). At the connection layer, two messages exist; SSH_MSG_CHANNEL_EOF and SSH_MSG_CHANNEL_CLOSE. SSH_MSG_CHANNEL_EOF is used when no more data is sent to a channel. This message has no response, unlike the other messages described. The channel remains open, and data can still be received on the channel.

When either end wishes to terminate a channel, SSH_MSG_CHANNEL_CLOSE messages are exchanged. When a host has sent and received an SSH_MSG_CHANNEL_CLOSE message, the channel is considered closed and the channel number can be reused. This message can be transmitted without having sent or received an SSH_MSG_CHANNEL_EOF message.

To terminate the connection at SSH's 'transport layer', the SSH_MSG_ DISCONNECT message is used. This has no response, as the sender stops sending and receiving after this message, and the receiver stops accepting data after receiving this message.

4.1.3 Advantages

Using SSH's port forwarding capability to create tunnels is relatively simple compared to other tunnelling methods. A secure tunnel can be created with the `-L` and `-R` options without requiring a reboot. SSH is available on all[3] Linux distributions, and is freely available as OpenSSH.[4] As the tunnels are defined not only by destination and source host but by port as well, traffic between the hosts can be limited to the specific tunnelled ports when used in conjunction with a firewall. The firewall prohibits other arbitrary, unsecured connections from being established between the two hosts. This would ensure secured traffic

[3]`http://www.openssh.com/users.html`
[4]`http://www.openssh.com`

is permitted traffic.

SSH can also be used to tunnel *all* traffic between two hosts, by tunnelling PPP over SSH. However, this is not considered an efficient VPN solution [24, 31].

4.1.4 Drawbacks

The architectural defects of SSH are investigated in [103]. One flaw identified is the key management mechanism. SSH has the concept of a certification authority (CA) to verify the authenticity of remote servers, however no widely deployed key infrastructure is available [125]. The host key of a server cannot be authenticated when a client connects to it for the first time, so clients cannot be sure if the server is authentic. When users connect to a remote server for the first time, they are presented with the following:

```
client:$ ssh server.company.com
The authenticity of host 'server.company.com (123.123.123.123)' can't
be established.
RSA key fingerprint is 86:e0:c0:f9:e8:2b:df:c6:1f:f4:bb:d9:13:2d:47:11.
Are you sure you want to continue connecting (yes/no)?  yes
Warning:  Permanently added 'server.company.com,123.123.123.123' (RSA)
to the list of known hosts.
user@client's password:
client:$
```

When the user types ‘yes’, the key is saved to `$HOME/.ssh/knownhosts`. On all subsequent connections, the fingerprint of the remote host is compared with the entry in the `knownhosts` file. It is up to the user to determine the authenticity of remote hosts; the `knownhosts` file is essentially the user's own CA [61]. Users can verify the authenticity of the host key via out of band methods (for example publishing the key fingerprints on a private webpage). Failing that, users can also compare the key fingerprint of the host after they have logged on, with the fingerprint presented. If the user was able to successfully connect to an intruder, however, it is likely their password has been compromised. Clients connecting to an intruder masquerading as a legitimate server form the basis of a ‘Man-in-the-Middle’ attack.

The option of not checking the remote server host key on the initial connection reduces the security of the protocol. Unfortunately this option is required as no key verification infrastructure exists, and the Secsh working group consider the ease of use critical to user acceptance of new security solutions.

The password user authentication method was also identified as vulnerable in [103], assuming clients connected to a fake server. SSH exchanges keys before the user is authenticated, so the intruder would be able obtain the user's password. This is because the password would have been encrypted using a session key negotiated between the client and the intruder. Again, this rests on the client being unable to verify whether the server was authentic in the first place, and connecting regardless.

The public key authentication method however is not vulnerable to this flaw. The server authenticates the user's public key with a valid user and checks the validity of the signature before authenticating the user. An intruder would not be able to carry out an attack as:

- The intruder's public key would not correspond to a valid user or;
- The signature would be computed over the client's username, public key and the negotiated session key between the client and the intruder, then encrypted with the client's private key. When using the victim's (client) public key, a valid signature would not be created, as the intruder must compute the signature over the session key between the intruder and the server, not the session key between the client and the intruder. This would then need to be encrypted with the client's private key, which the intruder does not have.

Finally a flaw in the authentication method negotiation was identified in [103]. If both the server and client cannot agree on an authentication method, the connection is closed. The server could then force the client to use an arbitrary method (such as the password method) of the server's choice if the user wanted to keep the connection.

Essentially these flaws rely on a user connecting to an intruder instead of the server. If users are vigilant and verify the fingerprint of the server's key before connecting the MITM attacks described would not be possible.

A disadvantage of using SSH tunnels is that only TCP traffic can be tunnelled securely; UDP applications cannot be used.

4.1.5 Implementations and Development

SSH is available from SSH Communications Security[5] as a commercial product. As a proposed internet standard, SSH requires at least one free implementation

[5] `http://www.ssh.com`

and this exists as OpenSSH.[6] OpenSSH is developed by the OpenBSD developers and has been part of OpenBSD since version 2.6. OpenBSD has a strict source code auditing process, ensuring the security of the operating system. Security problems are fully disclosed, enabling rapid fixes, usually within an hour.[7] Being part of OpenBSD, OpenSSH is subject to the rigorous auditing process as well. Both OpenSSH and OpenBSD are open source projects, and have the benefits as stated in Section 2.6.

4.1.6 Summary

SSH provides a secure method to administer remote systems. SSH can be used to create secured TCP tunnels between two hosts via port forwarding. This allows secure traffic for any TCP application. Establishing tunnels is simple and the OpenSSH implementation is freely available. As OpenSSH is open source, public code reviews ensure security flaws are rectified promptly.

Flaws in SSH rely on the user's inability to verify the authenticity of remote servers and consequently connecting to fake servers. Following this, if password authentication is used, the intruder can obtain the user's password. However, this can be prevented when public key authentication is used. MITM attacks are only possible when users connect and authenticate themselves to fake servers. Methods are available to verify the identity of a remote server before connecting, making MITM attacks preventable. SSH provides a simple, secure means of creating encrypted TCP channels and is a candidate tunnelling method.

4.2 IP Security (IPsec) - RFC 4301

IPsec defines a security architecture for the internet protocol in RFC 4301 [75]. IPsec is a widely deployed architecture for securing traffic over a public network and is well researched. Consequently, its features and capabilities are not introduced or described here. Section 3 of RFC 4301 [75] contains a sufficient overview.

Network Address Translation (NAT) can cause problems for IPsec [67]. When IPsec is operating in AH mode, the source address is hashed with other data for data integrity. When the packet passes through a NAT device, the source address will be changed. When the IPsec host receives this and validates its integrity, the resulting hash will be different and the packet will be blocked or dropped as it appears modified. IPsec can use ESP to traverse a NAT device,

[6] `http://www.openssh.com`

[7] `http://www.openbsd.org/security.html`

as the encapsulated source address is not changed by the NAT device; only the outer source address is modified.

4.2.1 Establishing a Security Association (SA)

The Internet Key Exchange Protocol version 2 (IKEv2) is the default protocol used by IPsec for automated key management and authentication. RFC 4306 [74] defines IKEv2. IKEv2 does not interoperate with IKEv1, however they both run over UDP port 500. This is actually an advantage, as it is impossible for IKEv2 to negotiate or roll back to IKEv1. This means flaws and vulnerabilities in IKEv1, which IKEv2 fixed, can no longer be exploited.

A Security Association (SA) is a unidirectional link secured by Authentication Header (AH) or Encapsulating Security Payload (ESP). To secure a connection, two SAs (one in each direction) are required and IKE creates SAs in pairs for this.

Messages are exchanged in pairs; a request and a response. The first two pairs of messages exchanged are the IKE_SA_INIT and IKE_AUTH messages. These four messages are enough to create the IKE_SA and the first CHILD_SA and are known as Phase 1 in IKEv1. A CHILD_SA is created through the IKE_SA, and represents a connection secured by AH or ESP.

Figure 4.9 illustrates the basic exchange. The initiator sends the first IKE_SA _INIT message. This message contains the cryptographic algorithms supported by the initiator, their Diffie-Hellman value and a nonce value. IKEv2 is required to support at least 3DES-CBC and AES-128-CBC for encryption, and HMAC-SHA-96 and AES-XCBC-MAC-96 for integrity [62]. IKEv2 may also support HMAC-MD5-96 for data integrity [106].

The responder replies with an IKE_SA_INIT message. This message contains the responder's choice of algorithms selected from the initiator's list as well as the responder's Diffie-Hellman value and nonce. This message may also request a certificate from the initiator. At this point, both sides can generate SKEYSEED from the nonces and the Diffie-Hellman values. All other keys are derived from SKEYSEED, and subsequent messages are encrypted and integrity protected.

The second pair of messages sent are IKE_AUTH messages. This exchange authenticates the previous two messages, authenticates identities and establishes the first (and often only) CHILD_SA. Identities are authenticated using one of three methods: a pre-shared key (PSK); a digital signature generated from a certificate issued by a CA; or a digital signature created from a self-generated public/private key pair. Each peer the initiator intends to communicate with must be configured with the PSK or public key when using a PSK or self-

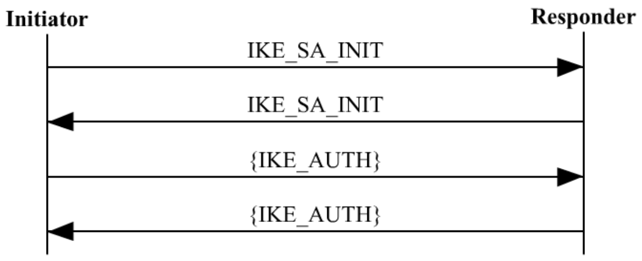


Figure 4.9: IKE_SA and CHILD_SA negotiation. These four message establish the IKE_SA and the first CHILD_SA. The first pair of messages negotiate algorithms and exchange nonces and Diffie-Hellman values. The IKE_AUTH messages are integrity protected and encrypted, as are subsequent messages (denoted by the {}).

generated key respectively. Consequently, these two methods do not scale well.

The initiator sends its authenticated identity, and 2 digital signatures; one for the current message (encrypted with the session MAC key), and one for the first message (encrypted with the private key). Traffic selectors[8] and an SA offer are also sent to start negotiation of the CHILD_SA. An SA offer lists the algorithm suites supported for the CHILD_SA. ESP is required to support at least 3DES-CBC, but may also support AES-128-CBC [49]. RFC 4305 requires DES-CBC not to be supported, as it is no longer secure. HMAC-SHA1-96 is the only required data integrity algorithm required by ESP, however AES-XCBC-MAC-96 and HMAC-MD5-96 are also supported. ESP also supports combined mode algorithms, which provide both encryption and data integrity. Combined mode algorithms can improve throughput and efficiency and may be required in ESP in the future. An example is AES-CCM; it is already used in WPA2/IEEE 802.11i [16] to secure wireless networks. AH has the same data integrity algorithm support as ESP.

The initial IKE_AUTH message may also contain the initiator's certificate and a list of trusted CAs - this is essentially a request for the responder's certificate. The digital signature used to verify integrity of the first message may be a private or shared key integrity code. The key used to generate the signature is associated with the identity of the sender (for example their private key).

The response contains the responder's identity (which is also authenticated by means of a pre-shared or a digital signature), a certificate (if required) and a

[8]These define the protocol carried in the SA, for example TCP, UDP and the start and end ports

digital signature over message 2 to assert its integrity. This second IKE_AUTH message also completes negotiation of a CHILD_SA, by choosing the set of sender supported algorithms. Signatures and MACs are verified, and this completes establishment of the IKE_SA and the first CHILD_SA.

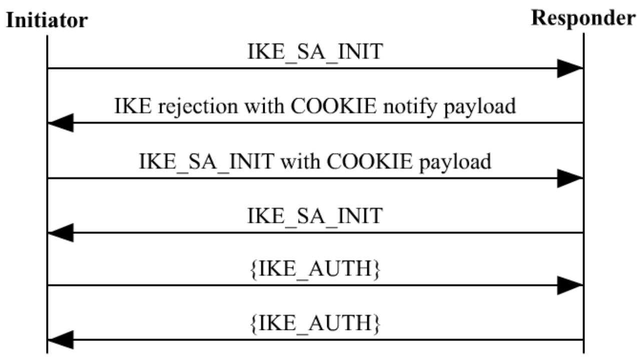

Initiator
Responder
IKE_SA_INIT
IKE rejection with COOKIE notify payload
IKE_SA_INIT with COOKIE payload
IKE_SA_INIT
{IKE_AUTH}
{IKE_AUTH}

Figure 4.10: IKE negotiation with cookies. This mechanism is invoked when the responder detects a large number of half open IKE_SAs. The initiator must prove it can receive from the address it claims to be sending from, before the responder allocates resources. After the first two messages, the rest of the exchange continues unchanged. Messages in {} denotes payloads that are encrypted and integrity protected.

A DoS attack is possible where the responder is flooded with SA initiation requests from forged IP addresses. The first response requires the responder derive a Diffie-Hellman key; this consumes CPU time and memory. When a responder is flooded with these requests in a DoS attack, legitimate clients are denied service. IKE uses cookies to protect against these types of DoS attacks. The concept is identical to the SYN-cookies used to secure TCP [118] and the DoS protection built into SCTP. Before the receiver commits any resources to the sender, it checks whether the sender can receive at the address it claims to be sending from. This defeats DoS attacks from forged IP addresses. Unlike SYN-cookies and SCTP's mechanism, which are always required, IKE only invokes the cookie mechanism when it detects a large number of half-open IKE_SAs (where the responder is waiting to receive the first IKE_AUTH message). Figure 4.10 shows the SA establishment when cookies are invoked. When cookies are required, responder rejects the initial IKE message (IKE_SA_INIT) unless it contains a COOKIE payload. This rejection message contains a COOKIE payload that must be included in future SA initiation requests. The initiator then resends the same IKE_SA_INIT, but this time includes the COOKIE payload, and the exchange continues the same as the standard exchange (Figure 4.9). When the responder receives the cookie, it knows the initiator's address has not been forged. Like the transport layer cookie mechanisms, cookies in IKE are computationally cheap to create and are stateless (no state is required to store them; they are simply recomputed). Another property of cookies is they cannot be forged; only the creator has the knowledge to construct a valid cookie, as it will be the creator verifying them.

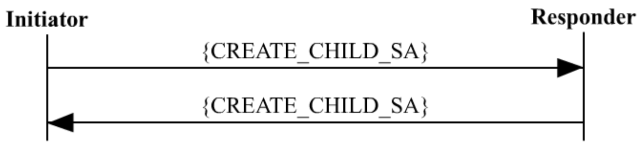


Figure 4.11: IKE creating a CHILD_SA. This may be initiated by either end of the IKE_SA after the initial exchanges. As these must take place after the IKE_SA negotiation, they are encrypted and integrity protected (denoted by the {}).

Additional CHILD_SAs can be created and this is referred to as Phase 2 in IKEv1. Like the IKE_AUTH messages, these messages are protected by the algorithms negotiated for the IKE_SA. Either end of the IKE_SA can initiate this exchange after the initial exchanges. Figure 4.11 illustrates this exchange, which is similar to the IKE_AUTH exchange. The CREATE_CHILD_SA request contains an SA offer, a nonce and proposed traffic selectors. The request may also contain a Diffie-Hellman value. The response contains the accepted SA offer, a nonce and traffic selectors. If a Diffie-Hellman value was in the request, a Diffie-Hellman value is included in response as well. Keys for the CHILD_SA can now be derived from the keys created during the establishment of the IKE_SA, the exchanged nonces and the output of the optional Diffie-Hellman exchange. This completes negotiation of the CHILD_SA.

The CHILD_SA exchange can also rekey the IKE_SA, in which case it would not contain the traffic selectors. Rekeying CHILD_SAs consists of creating a new CHILD_SA, then deleting the old one. Unlike IKEv1, IKEv2 relies on each end of the SA to initiate rekeying; IKEv1 negotiated an SA lifetime.

4.2.2 Deleting a Security Association (SA)

INFORMATIONAL exchanges are used to delete SAs, report error conditions and other administrative maintenance. They can also be used to check for liveness, which can be used in part for network reliability (Section 2.6). In

Figure 4.12, the requester sends an INFORMATIONAL message, listing the outgoing SAs to be closed. As SAs exist in pairs, the responder replies with the SAs to be closed in the other direction.

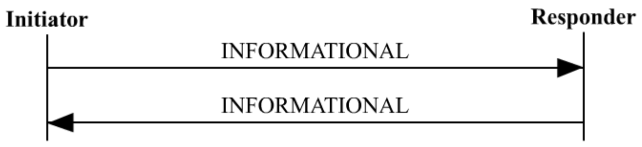


Figure 4.12: Deleting an SA. INFORMATIONAL messages contain control messages. When deleting an SA, they are closed in pairs. The responder closes the listed SAs in the INFORMATIONAL message, and the reply to the responder lists the SAs in the opposite direction to be closed.

4.2.3 Advantages

IPsec offers application independent security. Any protocol above IP is secured over any medium [1]. Applying security at a higher level protects only a single protocol, for example PGP only secures mail. Applying security at a lower layer protects only a single medium. IP is the most general protocol as it is common to all network stacks. Applications do not need to be modified to take advantage of IPsec, as it is often integrated into the operating system. This enables the application to function normally as the security is applied transparently.

Another benefit of IPsec compared to security applied at the higher levels is the packet drop performance [67]. IPsec blocks packets at a lower level in the network stack compared to higher level security protocols. This means relatively less processing is performed on the packet which can affect performance. This may have advantages during DoS attacks and heavy load scenarios.

IPsec is also very flexible; traffic passing through an IPsec client can be blocked, left unchanged or protected according to the SA on a host and port basis. This fine granularity enables a single SA to protect all traffic between two hosts or, conversely, each application or TCP connection can be protected by their own SA. As IPsec is designed for connecting networks, overhead is low when a single tunnel/SA is used to secure traffic between two security gateways. This adds to IPsec's scalability, as a single gateway and SA can serve many users.

IPsec is considered better than other IP Security protocols such as Microsoft's Point to Point Tunnelling Protocol (PPTP, RFC 2637 [57]) and Layer

2 Tunnelling Protocol (L2TP, RFC 2661 [120][9]) [51].

With PPTP, Microsoft Challenge Authentication Protocol (MS-CHAP, RFC 2433 [129]) authenticates the two endpoints of the tunnel and RC4 encrypts the payload as PPTP itself does not provide authentication or encryption [26]. However, an analysis of PPTP found several flaws [108]. Weak password hashing meant eavesdroppers could discover the user's password. Flaws in the authentication protocol meant an attacker could masquerade as the server. Flaws in the encryption implementation meant encrypted data could be recovered and keys generated from common passwords were breakable. Furthermore, attackers could use unauthenticated messages to crash PPTP servers. Microsoft resolved these issues with MS-CHAPv2 (RFC 2759 [128]) and Microsoft Point to Point Encryption (MPPE, RFC 3078 [88]). The security of PPTP however is still only as secure as the user's password, as MS-CHAPv2 is still vulnerable to offline password cracking. PPTP, however, can also be used with EAP-TLS, a robust and secure certificate based authentication method.

L2TP is often combined with the IPsec, as L2TP itself has no security functions and relies on IPsec to provide these [89]. L2TP's main advantage is it allows secure tunnelling of non-IP based protocols (when used in conjunction with IPsec).

4.2.4 Drawbacks

A cryptographic evaluation of IPsec is in [51], with the main criticism being IPsec's immense complexity. Complexity makes the system harder to configure or implement correctly, as well as increasing the probability of weaknesses and vulnerabilities. Security reviews on complex systems are also difficult, as the number of options make possible an exponentially large number of possible configurations to test.

IPsec's complexity is due to its development process. The ipsec working group[10] developed IPsec through a committee process. The "committee effect" (where the committee as a whole, makes poor decisions that no individual member of the committee would make themselves) manifests itself in IPsec as there are several ways of achieving the same thing. This unnecessary flexibility contributes to complexity, making it harder to test. IPsec's development process was compared to the committee process used by NIST, which organised a contest to develop the Advanced Encryption Standard (AES). NIST requested worldwide, algorithms meeting certain criteria, and selected one of the proposals

[9]The latest version of L2TP is L2TPv3, defined in RFC 3931

[10]The ipsec group concluded in April 2005

as the winner. This method is similar to the selection process used to procure equipment by the military.

The complexity of IPsec consequently results in complicated documentation. The working group produced over 30 RFCs, however the main RFCs are: 4301 (Security Architecture for the Internet Protocol), 4302 (IP Authentication Header), 4303 (IP Encapsulating Security Payload) and 4306 (Internet Key Exchange (IKEv2)). The IPsec documentation is also criticised in [51], as being incomplete (as they lack introductory material) and not specifying IPsec's objectives. The revised RFCs were released in December 2005 to address these issues, and in terms of readability, are an improvement (the evaluation in [51] was performed in 1999 using the RFC revisions which were produced in November 1998).

IPsec exists as an open standard so implementations from different vendors can interoperate. Despite this, most IPsec implementations do not interoperate well; both ends of the connection typically need to use IPsec implementations from the same vendor [22, 67]. This is because the IPsec documents allow for considerable leeway and variation concerning design choices and implementation details, resulting in IPsec compliant products from different vendors that do not interoperate. The December 2005 revisions improved on this somewhat, by specifying mandatory-to-implement algorithms. These are RFCs 4305 [49], 4307 [106] and 4308[11] [62]. At the time of writing, however, no interoperability tests or cryptographic analysis have been performed on the latest (December 2005) revisions of the IPsec standard.

Another disadvantage of IPsec is its close integration with the OS kernel [67]. This violates the secure OS Ring Architecture, where only the kernel and essential processes can run in Ring 0 (with the highest privilege), and other processes run in outer rings (which have a lower privilege, for example user processes run in Ring 3). Processes in outer rings cannot interfere with processes in the lower numbered, higher privilege inner rings. IPsec requires integration with the OS kernel (Ring 0) in order to have sufficient privileges to secure traffic. Operating with the highest privileges means if IPsec is compromised (due to an attack or a vulnerability exploited), the attacker will have full permissions to the system and have unlimited ability to damage the system. Requiring integration

[11]RFC 4308: "Cryptographic suites for IPsec" defines two cryptographic suites which are simply a collection of algorithms. For example, suite VPN-B is AES based, using AES for encryption and data integrity for both ESP and IKE exchanges and for the pseudo-random function. This simplifies IPsec as instead of having to negotiate the algorithms for each purpose separately (as in IKEv1), IKEv2 permits a suite to be chosen, which contains predefined algorithms for each purpose.

with the OS also makes installation difficult.

IPsec's fine granularity is also identified as a disadvantage in [51]. When an SA is created for each application or port (as opposed to using a single SA securing traffic on all ports), there is increased overhead, as many SAs must be negotiated. This drawback however is not limited to IPsec; it affects any VPN where each application has its own secure tunnel. Making more information available for traffic analysis is also mentioned as a drawback in [51].

While IPsec is designed to connect entire networks securely, its fine granularity enables secure tunnelling of a single application or port. Given IPsec's immense complexity, using it to tunnel a single port is excessive.

IPsec encrypts then performs the authentication on the ciphertext. This is also identified as a flaw in [51], as authentication should be applied to "what was meant, not what was said". In IPsec, this is not a serious flaw, as the encryption and authentication key are part of the same key in the SA. However, the ordering of operation (encrypting first and authenticating the ciphertext) makes it possible for a message to be authenticated as valid, however decrypted incorrectly.

Another drawback of IPsec is that it does not provide end-to-end application layer security [1]. IPsec only secures the link between hosts; data is not secured between users or applications. As IPsec operates on the network layer, traffic is encrypted and decrypted as it passes through the network interface. Hence, when a host receives encrypted data, IPsec decrypts it, then the plaintext is passed to the receiving process. Sending works the same; the application sends plaintext to the IPsec client running on the host. This is then encrypted before being sent out the network interface. Often IPsec is implemented in a security gateway to serve an entire LAN. A coarse end-to-end link is possible by giving each host its own tunnel to the local gateway, however this places additional load on the gateway. IPsec can also be implemented in the host's network stack (known as a Bump-in-the-stack implementation) to create an end-to-end link. However, in both these situations the traffic is still in plaintext *within* the host. Systems such as PGP can provide true end-to-end security; even within the host, the data is encrypted until the user explicitly decrypts it using a password.

Finally, IPsec does not perform user authentication [1]. IPsec only authenticates machines and relies on other independent mechanisms to authenticate users. The lack of user authentication and true end-to-end application layer security is a characteristic of IPsec providing security at the network layer. The network layer has no knowledge or concept of users.

4.2.5 Implementations and development

IPsec is ubiquitous; it has wide support in hardware devices (such as Cisco PIX, Juniper Netscreen and Watchguard Firebox appliances) and operating systems; Windows includes support and a free implementation exists for Linux as OpenS/WAN.

The third version of the IPsec documents were created in December 2005. These updates set to simplify, fix ambiguities and bugs as well as general improvement [60]. For example the IKEv2 document (RFC 4306) combined the Internet Domain of Interpretation (DOI, RFC 2407 [95]), the Internet Security Association and Key Management Protocol (ISAKMP, RFC 2408 [82]) and the Internet Key Exchange (IKE, RFC 2409 [59]) documents into one. RFC 4306 also contains elements concerning NAT traversal, legacy authentication and remote address acquisition from other documents. The main IPsec document, RFC 4301, also improved on readability (for example including Objectives and a System Overview section). The main documents also do not mention any algorithms; these are placed in separate documents (RFC 4305, 4307 and 4308). The ipsec working group recognised that algorithms may be superseded. By placing required algorithms in separate documents, when weaknesses are discovered in existing algorithms, or new ones designed, the main documents do not need to be modified; only the algorithm RFCs need to be updated.

4.2.6 Summary

IPsec is a mature security architecture for IP. IPsec operates at the network layer, providing security to any protocol operating above IP. This makes IPsec application independent. IKEv2 is used to perform mutual host authentication when creating Security Associations - a secured unidirectional connection.

IPsec offers better performance than security protocols operating at higher layers, as fewer resources are used to process packets (packets are discarded earlier). IPsec is flexible, so a single SA can secure an entire LAN, or each host or application can be given its own tunnel. This allows IPsec to offer the same granularity of security as other systems.

Unfortunately, IPsec suffers from system complexity as a result of the process used to standardise it. As a result, it spans several RFCs, however revisions have aimed to simplify and consolidate the documents. Improvements have also been made to develop the functionality.

Interoperability is another problem with IPsec. Even with standardisation, IPsec implementations from different vendors often do not interoperate. Again, development in the IPsec standard has aimed to rectify this.

IPsec also requires close integration with the operating system kernel. This makes the system vulnerable if IPsec is compromised. IPsec does not provide true end-to-end security; it can be approximated by giving each host its own tunnel, but within the host, data is still in plaintext. Finally, IPsec does not perform user authentication and relies on other mechanisms to provide this.

Despite these drawbacks, IPsec is a widely deployed architecture. Evaluations on IPsec have concluded that complexity harms its security. However, compared to other network level security protocols (PPTP, L2TP), IPsec is considered the 'least vulnerable', rather than the preferred method. Nevertheless, IPsec is capable of providing a secure connection and is a candidate tunnelling method.

4.3 Transport Layer Security (TLS) - RFC 4346

Transport Layer Security (TLS) is based on Secure Sockets Layer (SSL) 3.0, a security specification published by Netscape. The objectives of TLS are to provide cryptographic security, interoperability between independent applications, extensibility (in terms of incorporating new encryption algorithms) and efficiency in terms of processor requirements and network activity. TLS operates over reliable transport protocols such as TCP, and protects higher layer protocols such as HTTP (forming HTTPS). TLS secures traffic between two applications on a particular port. Netscape developed SSL to secure e-commerce transactions over the internet [76]. The SSL specification is developed into version 3 (SSLv3) and is supported in web browsers. This makes secure web transactions possible, with SSL the standard security for this. The IETF used SSLv3 as a basis to develop TLS. TLS is not interoperable with SSLv3, however it can roll back for compatibility with SSLv3.

TLS consists of two main components or layers: The TLS Record Protocol and the TLS Handshake Protocol. More accurately, it is the TLS record protocol that operates over a reliable transport protocol. The TLS record protocol provides encryption and data integrity for higher layer protocols. The TLS Handshake Protocol is one such higher layer protocol, and provides server and client authentication as well as secure negotiation of encryption algorithms and cryptographic keys.

The record layer protocol works by fragmenting the application layer data into blocks no larger than 2^{14} (16384) bytes (Figure 4.13). This is then optionally compressed according to the compression algorithm negotiated during

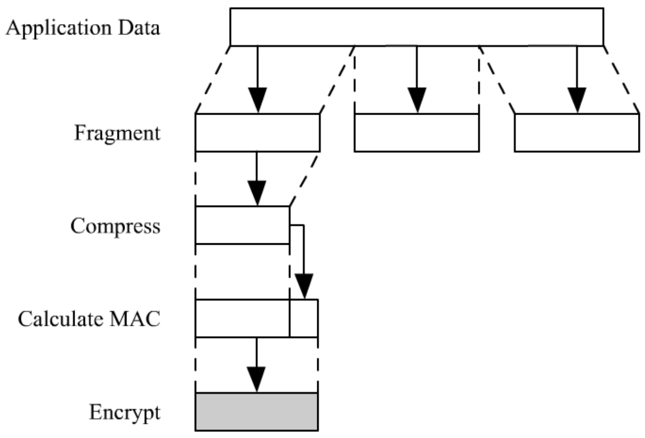


Figure 4.13: The TLS encryption process (from [110]). Compression is optional.

connection establishment. A MAC is calculated over the MAC key, the data fragment (that may be compressed), the sequence number, the fragment length, its type and the TLS version. This MAC is appended to the message fragment. The message is then encrypted. Instead of authenticating the ciphertext as IPsec does, TLS authenticates the plain text. Applying integrity protection to the plain text is preferable to protecting the ciphertext, as the ciphertext is dependent on encryption key used [51].

4.3.1 Connection Setup

TLS uses the TLS Handshake protocol to establish a connection. This requires the hosts to authenticate themselves (in at least one direction) along with instantiating and negotiating security parameters for the record layer. The TLS Handshake protocol has three sub protocols: the Handshake protocols, the Change Cipher Spec protocol and the Alert protocol.

The session is established with the Handshake protocol, shown in Figure 4.14. The client and the server exchange hello messages. The hello messages establish capabilities between the client and the server, such as the protocol version, session ID, cipher suite and the compression method. Each cipher suite defines a key exchange, encryption and MAC algorithm. The Client Hello message lists supported cipher suites, and the Server Hello contains the server's choice from the list. The messages also contain the nonces used for generating the key. The server follows with a certificate if it is to be authenticated. A server key exchange may also be sent if required; it is needed when the client is unable to transmit the pre-master secret via RSA public key encryption or

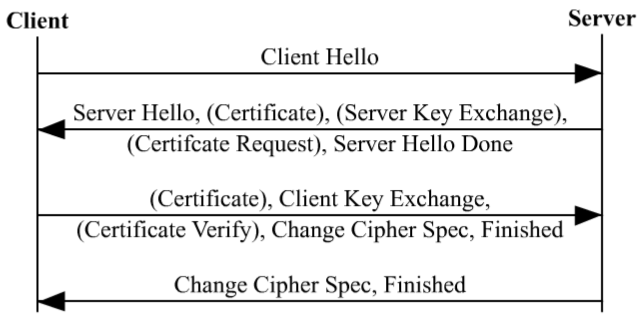

Client
Server
Client Hello
Server Hello, (Certificate), (Server Key Exchange),
(Certifcate Request), Server Hello Done
(Certificate), Client Key Exchange,
(Certificate Verify), Change Cipher Spec, Finished
Change Cipher Spec, Finished

Figure 4.14: TLS Handshake. The messages enclosed in brackets are optional.

a Diffie-Hellman exchange (for example if the server has no certificate, or if it is for signing only). The server can also request a certificate from the client. The server then sends the Server Hello Done message, and waits for the client's response

The client verifies the validity of the server's certificate (via a CA) and the server's chosen parameters. The client's response contains a certificate if requested and a Client Key Exchange message. This message sets the pre-master secret by sending an RSA encrypted secret (using the server's public key from the server's certificate or one provided in the ServerKeyExchange message) or by exchanging the Diffie-Hellman parameters, depending on the negotiated method. The master secret is generated from the pre-master secret. The master secret is used to generate a total of four values: a MAC secret and encryption key for the client and another set for the server.

If the client sent a certificate, a Certificate Verify Message is sent. This message is a digital signature of all messages starting from Client Hello up to but not including the Certificate Verify Message.

The Change Cipher Spec protocol is initiated at this point. This protocol is comprised of a single ChangeCipherSpec message. When a host receives this message, the newly negotiated cipher suites will be used for subsequent messages (that is, messages after this ChangeCipherSpec will be protected by the new algorithms). The ChangeCipherSpec message itself is protected under the current cipher suite, which is null or unprotected during the initial connection establishment.

The client then sends a Finished message immediately after the Change Cipher Spec message to verify the success of the key exchange and negotiation. This message contains a hash (using the session and MAC key) of all messages sent and received during the exchange up to this point. This message prohibits a downgrade attack by ensuring that the Client Hello's cipher list was not intercepted and modified such that only weak ciphers were available to establish a cryptographically weak connection. The Finished message is the first to be protected under the new algorithm suites, as it follows immediately from the Change Cipher Spec message.

The server sends its own Change Cipher Spec and Finished messages in response. These messages play the same role as they do in the client. The server's Finished message, however, includes the client's finished message in the hash. The key exchange and negotiation is considered successful if the client can validate the hash contained in the server's Finished message. At this point both sides have sent and received Finished messages. The handshake is now complete and application data can be exchanged.

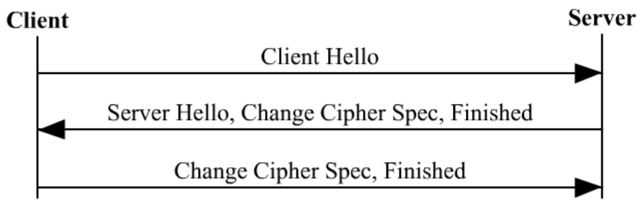

Client
Server
Client Hello
Server Hello, Change Cipher Spec, Finished
Change Cipher Spec, Finished

Figure 4.15: TLS Abbreviated Handshake. This flow can be used to resume or re-establish previous sessions, or rekey an existing session.

Client Hello messages are also used to rekey existing sessions, using the exchange illustrated in Figure 4.15. The Client Hello includes the session identifier of an existing session to rekey, as well as a new nonce. The server checks the session identifier for a match and if one is found, responds with a Server Hello with the same session identifier and an independently generated nonce value as well. Both the client and the server exchange Change Cipher Spec and Finished messages, and application data can then be exchanged. If the server cannot find a matching session identifier, a new one is generated and a full handshake is performed. The client can also use this abbreviated exchange to resume previous sessions and if the server is configured to re-establish old connections, the exchange that takes place is the same as that of rekeying.

RFC 4057 [104] defines an extension that enables the resumption of TLS sessions without state (such as session identifiers and cipher suites) stored on the server. The RFC lists four situations where stateless TLS session resumption would be useful:

- servers that handle a large number of transactions from different users

- servers that desire to cache sessions for a long time
- ability to load balance requests across servers
- embedded servers with little memory

The first three are applicable to this project; when a server must handle a large number of sessions from different clients, a lot of state is required which manifests itself through larger memory usage. This can lead to reduced performance. Sessions may be cached to avoid performing a full handshake each time a connection is established; this improves performance. Finally load balancing contributes to higher network availability and reliability.

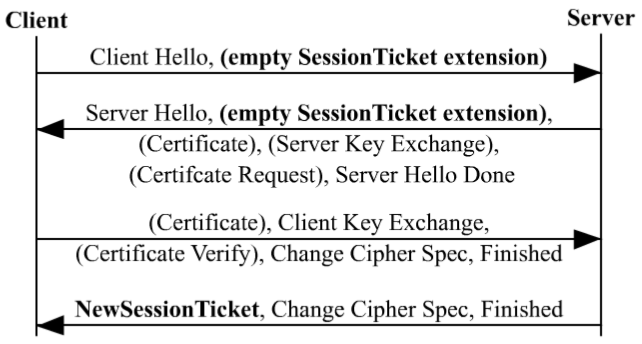

Client
Server
Client Hello, **(empty SessionTicket extension)**
Server Hello, **(empty SessionTicket extension)**, (Certificate), (Server Key Exchange), (Certifcate Request), Server Hello Done
(Certificate), Client Key Exchange, (Certificate Verify), Change Cipher Spec, Finished
NewSessionTicket, Change Cipher Spec, Finished

Figure 4.16: TLS Handshake with the SessionTicket extension. The NewSessionTicket message contains a ticket which stores the session state. This ticket is cached by the client, relieving the server of storing session state.

If this mechanism is supported by both the client and the server (by including an empty SessionTicket extension), the initial handshake includes a New Session Ticket message as indicated in Figure 4.16. The New Session Ticket message contains a ticket and is sent before the server's Change Cipher Spec message, after the client's Finished message is verified. A ticket is generated by the server and contains the session state (such as cipher suite and master secret). It is encrypted and integrity protected using a secret key known only by the server.

The client caches the ticket with other session parameters. When the client resumes the session, it includes the ticket in the SessionTicket extension in the Client Hello message (Figure 4.17). The server decrypts the ticket and verifies its validity. The server then resumes the session based on the state and parameters retrieved from the ticket. The server may also renew the ticket by including a NewSessionTicket message after the ServerHello. If the server chooses not to, the "empty SessionTicket extension" and NewSessionTicket messages are not included in Figure 4.17.

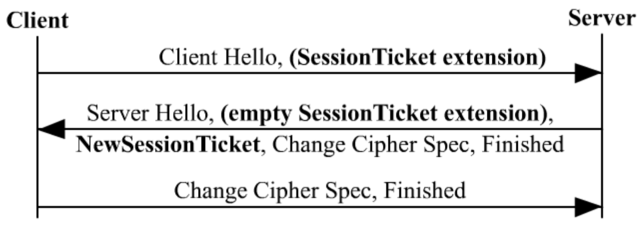

Client
Server
Client Hello, **(SessionTicket extension)**
Server Hello, **(empty SessionTicket extension)**,
NewSessionTicket, Change Cipher Spec, Finished
Change Cipher Spec, Finished

Figure 4.17: TLS Stateless Abbreviated Handshake. In this instance the server renews the session ticket.

If the ticket cannot be validated, the server performs a full handshake, and may issue a new ticket (Figure 4.16) except the "empty SessionTicket extension" message in the Client Hello contains the invalid ticket.

4.3.2 Connection Closing

The Alert protocol is used for describing various alerts, from handshake failure to illegal parameters, as well as the severity. Connections are closed using a close-notify message, carried using the Alert protocol. TLS uses the close-notify message to explicitly end a session and to avoid a truncation attack. SSLv2 closed the TCP connection to end the SSL session [2]. A truncation attack is possible in SSLv2, as attackers could forge TCP FINs to the recipient. The SSL connection would close as the recipient has no way of determining whether the TCP FIN was legitimate or not.

Either end of the connection can send a close-notify message. Once all pending data is transmitted, the close-notify message is sent, and the sender can close the write side of the connection. The receiver responds with a close-notify and closes the connection. The initiator of the close is not required to receive the responder's close-notify before closing the read side of the connection.

4.3.3 Implementations and Development

TLS is supported in all current web browsers, and has made e-commerce possible. TLS has been well tested and an inductive analysis of TLS concluded that it is a well designed protocol that meets its security goals [91]. TLS's maturity and its lack of known security weaknesses have made a widely used protocol.

The IETF's tls working group[12] was established in 1996 to standardise a transport layer security protocol. From SSLv3, the group published TLS [47]. The first revision of the TLS standard was in April 2006, updating TLS to TLS 1.1 [48]. The changes are small security improvements and clarifications. The tls working group plan to publish a TLS 1.2 revision, which removes TLS's dependency on the MD-5 and SHA-1 digest algorithms, includes new encryption modes and defines new cipher suites.

The following SSL-VPN and OpenVPN sections describe security architectures based on TLS.

SSL-VPN

An emerging trend offered by VPN vendors is the 'SSL-VPN'. A VPN securely connects entire networks together over a public, unsecured network. It enables traffic from any application, protocol or port to be transferred as if it was on the same private LAN. However, most 'SSL-VPN' products do not meet the above definition of a VPN at all.

Most products that claim to be an 'SSL-VPN' are marketed by the vendors as a clientless VPN solution. Vendors promote enabling employee access to the VPN via unknown clients, for example public airport or library kiosks [50], as the main benefit of an 'SSL-VPN'. There are several problems with this.

A popular understanding among IT publications [41, 50, 94] of the difference between IPsec and 'SSL-VPNs' is that IPsec a provides a secure layer 3 pipe to connect networks together or an extension of the network; the remote network can be treated as if it were on the same LAN. 'SSL-VPNs' are understood to provide secure access to a selected group of applications and services within the private LAN. Use of the term 'VPN' in this instance is incorrect, as a VPN provides secure tunnelling for any application.

Products declaring themselves as an 'SSL-VPN' fall under one of four categories: proxying, application translation, port forwarding or network extension [67]. Proxying is when an intermediary (such as an application layer gateway, ALG) is positioned between an internal and external application. The proxy is seen as the end point for both sides of the connection; it accepts client requests, rewrites them and forwards them to the server. Responses are returned the same way. The ALG can inspect the packet and examine the data to ensure it is well formed before forwarding the traffic. This additional processing by the proxy reduces the performance. The proxy does not support arbitrary protocols; each protocol supported requires separate coding.

[12] `http://www.ietf.org/html.charters/tls-charter.html` (Accessed October 2006)

Application Translation is when a protocol such as FTP is translated to HTTP and HTML for the display in a web browser. This can break the look and feel of applications as they are limited to the presentation capabilities of HTML. Like proxying, this method cannot support arbitrary protocols; each protocol that is to be supported requires creating a translator.

Port forwarding securely forwards traffic to a specified destination based on the port (for example SSH's port forwarding, described in Section 4.1). Like the other two methods, this does not support arbitrary applications, as each application requires forwarding of its own port. This requires client software on the host client to forward the traffic.

Network extension is the only method that provides a true VPN. Like IPsec, client software is required on the host to create a secure tunnel. This tunnel supports any traffic on any port, and is the only method that meets the definition of a VPN.

Vendors claim clientless access by using SSL/TLS enabled web browsers. This only works with proxying and application translation. Even then, it only works well with browser based applications. Applications that cannot be displayed using HTML typically require a 'desktop agent' or a client downloaded to the client host in order to be presented properly [94]. These small Java or ActiveX clients can also be used to secure SSL unaware applications [41]. Thus the vendor's claim of providing clientless access is misleading, as these small thin clients encapsulate application traffic through secure tunnels.

Vendors claim the clientless solution gives remote employees access form anywhere. This itself has several problems. The web browser's security policy may prohibit Java applets, ActiveX controls, plugins or other active content [94]. This means browsers may need to be reconfigured to allow these, otherwise these thin clients would be blocked. However, this reduces the security as Java and ActiveX controls are applications, much like a normal executable. The Java applet or ActiveX code from other sites may contain instructions to download and execute software, all without the user's knowledge, permission or intervention. Because these are applications, they have a high potential to cause damage, especially when the user does not know they are being executed.

A second more serious issue is that the remote employee can connect from any public client. This carries a huge risk, as public hosts are by definition untrusted and uncontrolled. Public hosts (such as airport kiosks) may be infected with trojans, viruses, worms and other malware [41]. An active attack can take place, for example a trojan can start a worm attack on the network through the secure tunnel. A passive attack can also take place in the form of a keystroke logger; when the remote employees authenticate themselves to the network with a username and password, the keystroke logger would have recorded this, nullifying the credentials. Networks are only as secure as the devices connected to it.

Another problem with using public uncontrolled hosts is that session state is left behind after the session has ended such as cached credentials, browser history, cookies and temporary files. Many SSL-VPN solutions address this by using applets to delete this state [94]. These applets also mitigate risk by checking the capabilities of the public host, for example ensuring antivirus protection is present. The applets can also use access rules to restrict permissions and limit the functions available when the VPN is accessed from a public host, for example restricting uploads. However, administrative access to the machine is required to carry out these tasks [67]. If administrative permissions are available on a public machine, it is likely to already contain keystroke loggers, remote control software and other malware.

Finally the 'SSL-VPN' solutions are proprietary. Section 2.6 discusses why proprietary protocols are rarely better than public open standards. An application or applet developed for one 'SSL-VPN' product, such as Juniper's Netscreen, is unlikely to work on another, such as AEP Networks' Netilla Security Platform, without modification.

OpenVPN

OpenVPN[13] is an implementation providing a true VPN via network extension using TLS. OpenVPN uses the OpenSSL toolkit[14] for cryptography. Both projects are open source.

OpenSSL implements SSLv2/3 and TLS 1 as well as a general purpose cryptography library. The TLS mode of OpenSSL was certified by the National Institute of Standards and Technology (NIST) as FIPS 140-2 compliant in January 2006 [70]. The Federal Information Processing Standard (FIPS) 140-2 defines security requirements for cryptographic modules. OpenSSL met the Level 1 requirements under the Cryptographic Module Validation Program (CMVP). U.S. Government agencies and regulated industries (such as finance and health care) are required to use FIPS compliant products to store, collect and transfer data that is 'sensitive but unclassified' [13]. In July 2006, however, the certificate (#642) was revoked [69]. It is still available from the Computer Security

[13] `http://openvpn.net`
[14] `http://www.openssl.org`

Resource Centre (CSRC) Validated Modules page.[15]

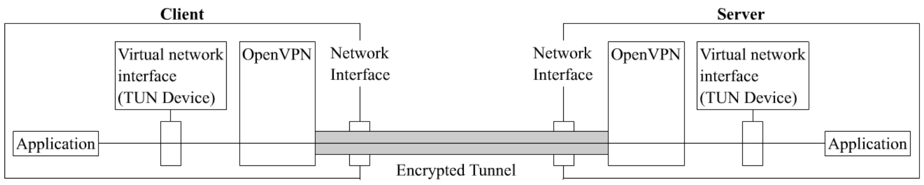


Figure 4.18: OpenVPN and a 'TUN' device. Applications see the virtual 'TUN' device as a network interface. Traffic sent to the virtual 'TUN' device is forwarded to the OpnVPN service. OpenVPN then encrypts the traffic before sending it out the network interface.

OpenVPN uses virtual networking interfaces to create encrypted tunnels. A 'TUN'[16] device is a virtual network adapter that appears as a point-to-point device to the operating system [7, 127]. Data sent to an ethernet device is sent out the wire by the ethernet driver, whereas data sent to a TUN device is sent to user space applications. Applications can open the TUN device like a folder or socket, and can read from and write IP packets to it. A 'TAP'[17] device is also a virtual adapter, but simulates an ethernet device. Figure 4.18 illustrates how a TUN device is used with OpenVPN. Data is sent to the TUN device, which is forwarded to OpenVPN. OpenVPN uses the OpenSSL library to encrypt the data, and then sends it out the real network device. Decryption is the opposite; OpenVPN reads data from the real network device and decrypts it before sending it back to the TUN device. The application then reads the data from the TUN device.

OpenVPN encapsulates the IP data into a UDP datagram before sending it out the real network interface. When TCP data in an IP packet is encapsulated back into TCP (Figure 4.19), the traffic reliability mechanisms provided by TCP are duplicated, for example retransmissions and acknowledgements. This reduces efficiency, as the lower TCP layer guarantees delivery, so the higher TCP layer does not need retransmissions. TCP's retransmission algorithm uses adaptive timeouts; when an ACK is not received after the timeout period, the sender retransmits the packet and increases the timeout. This behaviour lets TCP operate on connections with varying bandwidth, latency and packet loss

[15]`http://csrc.nist.gov/cryptval/140-1/1401val2006.htm`

[16]TUN does not appear to be an abbreviation, but is accepted to mean a virtual point-to-point network device.

[17]Like TUN, TAP also does not appear to be an abbreviation, but refers to a virtual ethernet network device.

rate.

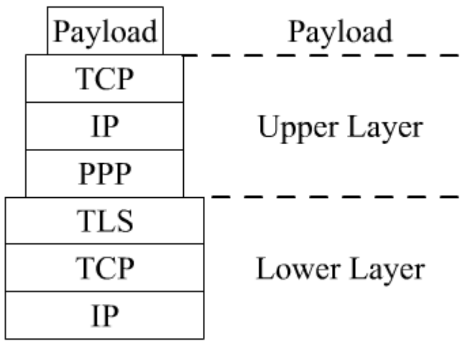


Figure 4.19: When TCP is encapsulated back into TCP, the upper layer is not required to send retransmissions or perform other reliability functions, as the lower layer ensures this.

The two TCP layers in Figure 4.19 have independent retransmission timers. As described in [119], if the actual connection loses packets, the lower TCP layer will retransmit, and increase the retransmission timeout. The upper TCP layer will not have received an ACK, so it too will send a retransmission and increase its own retransmission timeout. The retransmission timeout of the upper layer will always be less than that of the lower layer, however, as the increases began slightly after. Consequently the upper layer TCP will queue retransmissions faster than the lower layer can process them. Hence OpenVPN encapsulates data into UDP; TCP expects an unreliable carrier and UDP provides a carrier close to the native IP environment.

4.3.4 Advantages

This section and the following Drawbacks section refers to OpenVPN and the TLS protocol, not ‘SSL-VPNs’. OpenVPN runs in user-space. This makes OpenVPN more secure than VPN products that are closely integrated with the OS, for example IPsec. OpenVPN conforms to the secure OS Ring Architecture as explained in Section 4.2.4. OpenVPN uses the virtual network device, instead of the actual network device, which requires low level access. As low level access is not required, OpenVPN does not require integration with the OS kernel. OpenVPN runs as a user process (Ring 3) and this simplifies installation, configuration and portability; OpenVPN can run alongside IPsec without any conflicts [67]. Because OpenVPN runs as a user process, its permissions can be restricted to that of a limited user. This means if the OpenVPN service is compromised, the attacker will only be able to perform the same actions as a limited user. Another benefit of running as a limited user (opposed to an OS kernel process) is that OpenVPN can be 'constrained' to a specified directory. This 'containment' denies OpenVPN access to the rest of the system. This layering of protection (defence in depth) protects the system if OpenVPN is compromised, as the attacker's potential to damage the system will be extremely limited.

OpenVPN encapsulates IP packets into UDP datagrams. Overall, UDP based VPNs add less overhead, have better bandwidth utilisation and have a lower latency than TCP based VPNs [76]. OpenVPN can also be configured for load balancing using IPtables. Rules can be created to share the traffic between a range of addresses, transparently to the applications [67].

As OpenVPN provides a secure VPN tunnel similar to IPsec, applications can be secured without modification. This is because the TUN virtual device appears and functions as a normal network adapter. The OpenVPN architecture is analogous to a Bump-in-the-Stack (BITS) or a Bump-in-the-Wire (BITW) IPsec implementation, and therefore offers similar functionality.

TLS can perform user authentication as TLS operates at a higher layer than IPsec. Servers can authenticate users by requesting a certificate, however this is rarely used in e-commerce sites as no Public Key Infrastructure (PKI) exists to validate users (PKI is currently used to only authenticate servers in an e-commerce environment). TLS, however, cannot perform user authorisation, and relies on the upper layers.

End-to-end security is possible with TLS, as TLS encrypts at the application layer. Applications that are natively TLS enabled benefit from end-to-end security as data is only decrypted when it reaches the receiving application. However, when OpenVPN is used, the connection is no longer strictly end-to-end. Encrypted data passes through the network interface to OpenVPN, which decrypts it, and forwards it to the virtual TUN device. Any application listening on the TUN device (for example ethereal or wireshark) will see the data in plaintext. Applications listening on the actual network interface, however, will still see the ciphertext as OpenVPN has not decrypted it yet.

TLS connections can successfully traverse NAT devices. NAT devices typically change source addresses and ports. TLS is not affected, as the MAC is calculated over the data, the MAC key and a sequence number; the MAC does not include the address and ports.

TLS is also application independent as higher layer protocols can be layered on top (for example HTTPS for HTTP and SIPS for SIP URIs). OpenVPN extends this so that applications do not need to be aware they are using an encrypted tunnel. TLS's only requirement is that it operates over a reliable transport protocol.

Finally hardware accelerated SSL/TLS devices exist to offload the computationally expensive cryptographic functions from the main CPU. OpenSSL can be accelerated with these devices, and OpenVPN can benefit as well as it uses OpenSSL for its cryptography functions.

4.3.5 Drawbacks

As TLS operates at a relatively high level, the packet drop performance will be lower than security protocols that operate at lower layers (for example IPsec). Packets are processed by more layers in the network stack before they are discarded. However, hardware accelerators can be used to improve overall TLS performance.

Another drawback is client authentication. As no PKI exists for clients, e-commerce websites only authenticate the server with certificates, and authenticate the user with a username and password. In a VPN this is less of a problem, however, as users can obtain certificates from their own corporate CA. For the purposes of this project, this is not a problem as the remote devices (instead of users) would preloaded with certificates distributed by the corporate CA (for example the trusted manufacturer of the transaction terminal).

4.3.6 Summary

TLS is based on SSLv3. It is widely deployed and developed and is the standard way to secure e-commerce transactions over the internet.

This section described three systems: 'SSL-VPNs', OpenVPN and TLS/OpenSSL. 'SSL-VPNs' are generally products that use the term VPN incorrectly. These products are usually nothing more than proxies or application translators and fall far short of the definition of a VPN. The advantage 'SSL-VPNs' have over other VPN products is that no client software is required. However, when port forwarding or network extension is the method used, clients are required. 'SSL-VPNs' have flaws in their security model and are also proprietary.

OpenVPN is an open source VPN product that offers application/port independent tunnelling functionality similar to IPsec. However, OpenVPN runs in user space. Running in user space gives OpenVPN several advantages over IPsec, such as ease of installation and configuration as well as being able to contain the damage by an attacker if compromised. OpenVPN uses OpenSSL, an open source cryptographic library that implements TLS.

TLS is capable of authenticating users via certificates however this is rarely used in e-commerce due to a lack of an appropriate PKI. While TLS may have lower performance than other systems due to operating at a higher layer, performance can be accelerated with specialised hardware.

TLS is still under active development. These continued improvements will ensure the security of e-commerce. TLS's advantages and lack of drawbacks make it candidate tunnelling method.

Chapter 5

Architectural Analysis

A transaction network can operate over two contrasting network models: the open Internet model and the controlled IP PSTN model. This section discusses the merits of each and the challenges they bring.

5.1 Open Internet

In this model, access is via an Internet Service Provider (ISP, Figure 5.1). Clients are generally registered only with the ISP and are otherwise anonymous on the Internet; clients have no fixed identifier when connecting with a dynamic IP address.

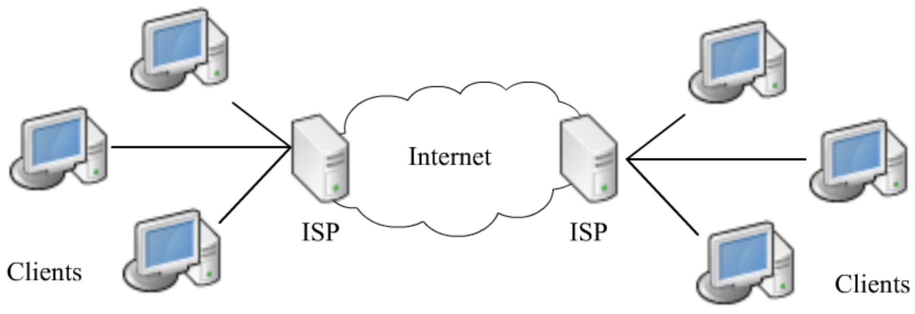


Figure 5.1: The Open Internet model. Clients access the Internet via an ISP and are otherwise anonymous.

Once a client is directly connected, it is by default exposed to every other client on the Internet. Conversely a client can attempt to access any other connected client in the world, as there is no access control or logging other than those performed by the ISP for billing purposes. This unregulated environment brings risk as all types of network attacks are possible, such as intrusion, information theft and DoS attacks.

Firewalls can be implemented at the endpoints to reduce these risks and NAT is often used to conceal internal network infrastructure information. NAT, however, often causes connectivity problems in applications when the application inserts private non-routable addresses in the packet payload. NAT devices also remove the end to end significance of the IP addresses of the hosts on the two ends of the connection. This poses a problem for communications that rely on peer to peer connections such as VoIP and file sharing applications. Hosts behind a NAT device cannot receive unsolicited incoming connections; hosts behind a NAT must initiate the connection. Several solutions exist to work around this, such as Universal Plug and Play (UPnP) and Simple Traversal of UDP through NATs (STUN). However, these solutions are not scalable or widely implemented [18].

5.1.1 Advantages

The advantage of an open peer to peer model is not having a single point of failure; for example, in peer to peer file sharing networks, nodes are both clients and servers. However, a transaction network does not benefit from this advantage as it uses a client-server model as noted in Section 2.6.

The Internet is also generally fault tolerant and when paired with a presence architecture (for example SIMPLE, used in IM applications), real time knowledge of the status of network components is possible. This can contribute to network availability as clients will know which servers are available, and vice versa if required (transaction servers typically do not need to know the status of transaction clients).

5.1.2 Drawbacks

There are several drawbacks to operating a transaction network over the Internet. A major disadvantage is the lack of security; in an uncontrolled environment, data can be monitored as it passes through intermediaries. Encryption and data integrity protection largely protect from this risk however.

As every directly connected device is exposed to every other device, DoS attacks on transaction processing servers are possible. Attackers are anonymous and untraceable as source IP addresses can be spoofed. Network infrastructure can be protected from DoS attacks by using class based queuing and real time traffic monitoring as detailed in [73]. Such measures can also contribute to scalability; a load balancer placed between the transaction servers and the Internet can tunnel transactions to the transaction processors with the least number of connections. The load balancer can also use the class based queuing mechanism to throttle transactions during congestion or peak load situations.

The Internet model also lacks QoS guarantees; it is typically a best effort service. A transaction network requires a low latency, reliable network. These parameters cannot be guaranteed in the open Internet model.

As the Internet is uncontrolled, real time network management functions are limited to the server's side; successful transactions can be measured on the server only. Unsuccessful transactions, however, cannot be measured in real time without co-operation with the client's ISP. Statistics on failed transactions initiated by the client can however still be reported 'offline', for example at the end of each hour or day.[1] However, other management functions such as knowing which transaction servers are congested or offline are still available, by using a presence architecture such as SIMPLE.

5.2 Controlled IP PSTN

The term "IP PSTN" refers to a private IP network desgined for VoIP, rather than a public switched telephone network. In this model, access is via an IP PSTN provider (Figure 5.2). Clients reside on their own subnet and access the private network through the provider's Session Border Controller (SBC). SBCs are also known as back-to-back user agents (B2BUA) and are essentially an application layer gateway and proxy and are situated between the participants of a session. They control access to the private network and this gives the provider complete control over policies for admission to the service.

Each client has a common identifier (a SIP URI) and must register with a SIP registrar before accessing the network. This assists in NAT traversal, as the SBC becomes the source and destination for all signalling messages and media streams entering and leaving the provider's network [15]. SBCs can be considered an implementation of the mechanism defined in [100].

When a client registers with a SIP registrar/call agent (Figure 3.22), the SBC maps the source IP address and port of the REGISTER message to the client's SIP URI. The IP address and port mapped however do not actually belong to the client; they belong to the client's firewall/NAT device. The SBC

[1]Offline here means "not in real time" (that is, not reporting failure as it occurs), rather than being disconnected from the Internet.

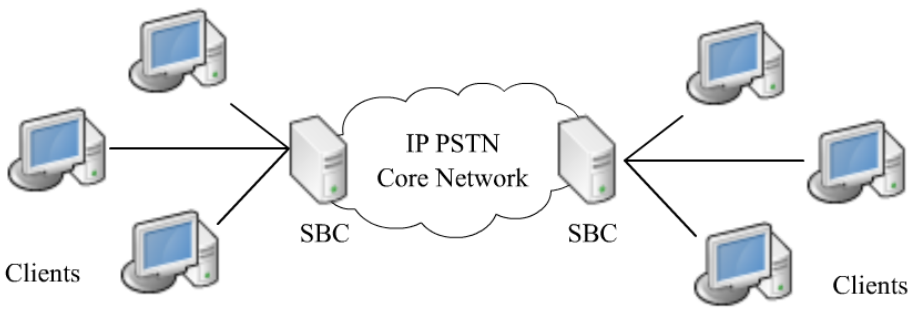


Figure 5.2: The Controlled IP PSTN model. Clients (which may be IP telephones) connect to the IP PSTN Core network via Session Border Controllers (SBC), which are under the control of the service provider.

modifies the REGISTER message so the CONTACT and VIA fields point to the SBC [18]. The SIP registrar then binds the client (via the SIP URI) to the SBC's address.

When the call agent is queried on the location of the client (Figure 3.20), the SBC's address is returned. Hence signalling messages addressed to the client are directed to the SBC. When the SBC receives a signalling message addressed to the client (for example an INVITE), it is forwarded to the IP address and port associated with that client's URI.

When the client initiates a call, the INVITE request is sent to the SBC. The SBC replaces any relevant fields in the INVITE request such that it appears to originate from the SBC. The modified INVITE is then forwarded to the call agent. The response will go back to the SBC, which will forward it to the client behind the firewall/NAT device. This makes the SBC the source and destination for all sessions.

Media streams are also proxied through the SBC. The SBC knows which IP and port the firewall/NAT device allocated for the client's outgoing media. This IP address and port is associated with the client's URI. The SBC then forwards incoming media addressed to the client's URI to the same IP address and port. By performing application layer gateway functionality (by modifying the fields in SIP messages) and proxying (by making itself the source and destination of all messages), SBC's can successfully traverse NAT devices without modifications to existing hardware, software, firewall configurations or security policies.

5.2.1 Advantages

There are other benefits to having a provider controlled SBC situated in the signalling and media path. SBCs have the topology hiding security benefits of NAT, as the SBC conceals the details of the client's internal network topology. Clients receiving session invites will always see the SBC as the source, not the originating network or client. Outgoing calls are proxied through the SBC, which replaces the real source address with its own.

Clients can only access other clients permitted by the SBC, unlike the Internet model, where any other client is accessible by any other. As clients must initiate sessions through the SBC, the SBC can deny any session invite via policy if required. The SBC can also completely deny access to unauthorised clients. SBCs enable service providers to provide a VoIP service, while replicating the traditional phone networks. Service providers can retain their existing business models in a VoIP network as they have full control over the network.

The SBC can also protect network and service infrastructure from attack. As the client's real address is not exposed, it is protected from port scans and DoS attacks. Conversely a client cannot initiate port scans or DoS attacks on other clients, as the SBC will not allow the traffic onto the core network.

SBCs also enable the service provider to manage and control sessions. This can be in the form of logging failed transactions in real time for example. As all traffic must pass through the SBC, it can also throttle traffic if it is abnormally excessive (for example during a DoS attack). This may also happen during peak load scenarios; the throttling mechanism enables clean failure. Transaction processors will gracefully reject new requests, but continue to service current requests. This highly regulated private network environment also makes QoS possible; SBC's can provide a link with a guaranteed level of service (for example minimum latency).

5.2.2 Disadvantages

Operating a transaction network over an IP PSTN also has disadvantages. The SBC, with all the benefits it brings to an IP PSTN is also the source of all the drawbacks in this model. The SBC represents a single point of failure, as all connections and sessions run through it. Having multiple redundant SBCs reduces this risk.

As all connections access the IP PSTN through the SBC, it can become a bottleneck, limiting performance and security. The SBC breaks the end to end transparency as it acts as the called client, then places a second call to the actual destination. This can introduce latency. Ensuring sufficient SBCs are deployed to handle peak, not average load scenarios can reduce these drawbacks.

The capabilities of clients are also limited to the capabilities of the SBC, as the SBC functions as a proxy. This may not be a problem in a regulated transaction network, as all clients will be uniform in capability and specification.

Control of the session is mediated by the SBC. This is a drawback if the clients do not trust the service provider. End to end encryption cannot be used, as SBCs need to be able to modify packet contents. Encryption of the payload, however, is possible as long as the headers the SBC modifies is in plaintext, or the SBC has the ability to decrypt the headers.

5.3 Conclusion

The open Internet model offers clients relatively anonymous and unregulated access to every other client connected. Clients are also exposed to every other connected client. This brings security risks. Firewalls can be used to reduce the risk, however this introduces NAT traversal issues. While solutions exist, they are not scalable or widely implemented. Even if an elegant solution to NAT traversal was developed, the Internet model lacks QoS abilities, a requirement for a transaction network.

The IP PSTN however offers a highly regulated QoS enabled service. The high level of control wielded by the service provider (via the SBCs) ensures security, as clients can only access other clients allowed by the SBC. As the SBC's separate the clients by acting as a proxy, no network topology information is revealed. This protects against DoS attacks. SBCs can also throttle traffic, as they are QoS enabled. This also protects against DoS attacks, but also enables the network to function during peak load scenarios; the transaction processing rate will drop, but not fail completely.

Some disadvantages of running a transaction network over an IP PSTN can be overcome by ensuring enough SBCs are deployed, such that they do not become a potential bottleneck or present a single point of failure. Security also relies on the clients trusting the service provider and SBCs. Table 5.1 summarises the difference between the two models.

Overall the advantages of an IP PSTN over the Internet make it the preferable network architecture on which to operate a transaction network.

Property	Open Internet	IP PSTN
Access	Public via ISP	Controlled via SBC
Connection	Direct connection	Proxied via SBC
Client Anonymous?	Yes	No
Client Accessible to Others?	Yes	No unless permitted by SBC
Client can Access Others?	Yes	No unless permitted by SBC
QoS Possible?	No	Yes
Network Management Functions?	Partial	Yes
End to End transparency?	Yes	No
NAT Traversal?	Needs additional server	Yes

Table 5.1: Summary of differences between the open Internet model and an IP PSTN

5.4 Architectural Options

This section describes architectures built on the two different models, and how they would meet the requirements.

5.4.1 IP PSTN Solution

A solution based on the IP PSTN could be arranged as in Figure 5.3. Clients access the core network via an SBC. The SBC forwards the transaction from the client to the receiver's SBC (SBC2). The receiver in this case is a transaction processor. There may be a concentrator physically placed between the transaction processor and the SBC as illustrated, however the concentrator and transaction processor can be logically considered a single component. A concentrator combines transactions from many connections into a single connection. The concentrator also operates as a firewall, protecting the transaction processor from harmful traffic. Multiple transaction processors may be positioned behind the concentrator for load balancing, for example, the concentrator may allocate transactions on a round robin basis.

SIP would be required to access the core network via the SBC. The SIMPLE extensions could also be used, giving the network a degree of service awareness (for example notification of which servers are unavailable, offline or congested, described in Section 3.6.4).

A total VPN solution providing transparent network access (using the remote network as if it were a local network) is not required in this situation; clients send transactions to the transaction processor and receive acknowledgements.

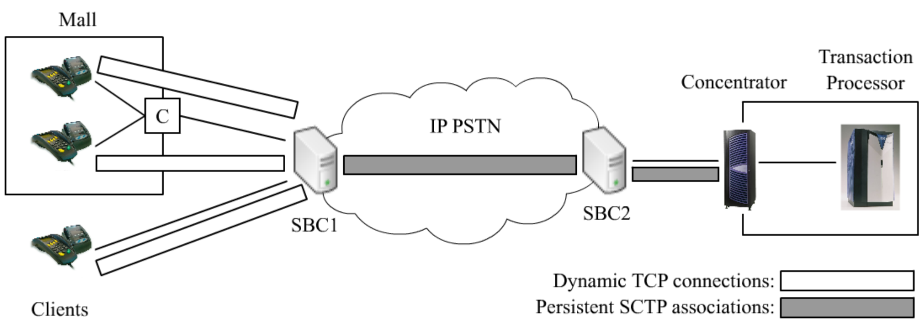


Figure 5.3: One possible IP PSTN based solution. The connections between clients and SBCs would be dynamic, while the connection between trusted infrastructure (SBCs, trusted concentrators) could be persistent SCTP associations. Concentrator C is untrusted and cannot decrypt the data to aggregate traffic. The Concentrator in front of the Transaction Processor and the Transaction Processor itself can be considered a single logical unit.

Consequently, the VPN candidates (IPsec and TLS based VPNs) described in Section 4 are excessive when the network requires only a single port to be tunnelled or secured.

SSH port forwarding and TLS are both protocols for securing a single port. However, SSH port forwarding assumes a client can log into the SBC to establish the receive end of the SSH tunnel. Even when using public/private key authentication, permitting login access to the SBC exposes it to unnecessary risks. Once logged in, an intruder could execute malicious code, or induce denial of service by rebooting the SBC (assuming the intruder had overcome any restrictions to administrative access and execute permissions). Other DoS attacks against SSH are described in [56].

TLS then becomes the preferred security protocol, as it does not require the SBC to allow login by clients. Authentication would be two-way via public/private key authentication. In the case of EFTPOS, a trusted vendor issues the clients (EFTPOS terminals) and would therefore issue them with a certificate at the time of manufacture. This avoids the client authentication problem present in e-commerce due to the lack of client side PKI. The TLS tunnels must terminate and begin at the SBCs, as SBCs must be able to modify packet contents as described in Section 5.2.2.

One issue here is whether or not a single manufacturer issues the same certificate to all the clients (such as EFTPOS terminals) it manufactures. If a single certificate is used and the key is broken or the terminal is somehow compromised, the entire system would also be compromised. Global secrets are a bad design feature, and once compromised, are difficult to recover from [107]. The contrary requires the manufacturer to install different certificates/keys/secrets in each terminal.

This raises another related issue of whether the terminals can be modified or upgraded. If they can be upgraded, then in the event of key compromise, they can be installed with a new secret key. However, this also enables tampering of the terminals; it could be possible for malicious third parties to upload their own certificate or secret into the terminal. Conversely, if the terminals were made tamperproof and not upgradeable, it would not be possible for malicious parties to read the contents of its secret, nor install their own keys. Obviously, in this situation, manufacturers would not be able to upgrade or install new secrets. One solution could be to have terminals that are upgradeable, but unreadable. The terminals would have to be tamperproof to avoid malicious interference.

While these last two client terminal issues (global secrets and upgradeability) are important, they stray beyond the scope of this project.

The connection could operate over either TCP or SCTP. While TCP is stream based, SIP's SIMPLE (more specifically the MESSAGE function) provides the application layer framing to distinguish between separate messages. Both are reliable transport protocols with flow control mechanisms.

When a client requires access to the core network, it must register with the SBC (as described in Section 5.2) to support NAT traversal. This would involve the following steps:

1. TCP/SCTP establishment
2. TLS establishment
3. SIP REGISTER
4. SIP SUBSCRIBE (optional; this would let the client know the status of the transaction processors)
5. TLS close notify
6. TCP/SCTP close

After a client has registered with the SBC, it may send transactions. One issue is whether clients should use a single secured SIP session for every transaction, or establish and close a secured SIP session each time a transaction is sent (Figure 5.4).

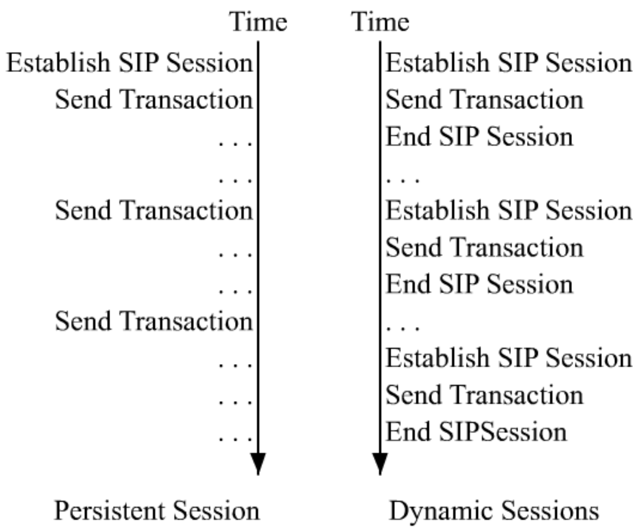


Figure 5.4: With a persistent session, the session is established and every transaction uses the same session. This avoids the overhead of establishing and closing a session for each individual transaction. Dynamic sessions mean every transaction gets its own session. This frees resources on the receive end, which may otherwise need to maintain multiple idle sessions.

The IP PSTN model replicates the operation of the traditional telephone networks, for example in Figure 3.23, the 180 Ringing message may be conveyed to the caller using an IP telephone as a ringing tone. Current EFTPOS terminals dial into the telephone exchange each time a transaction is sent; the telephone exchange then re-routes the call to a Network Access Controller. After the transaction is sent, the EFTPOS terminal hangs up the connection and relinquishes the line.

Accordingly, transaction terminals in the IP PSTN model would use new sessions for each transaction as a result of using a network which replicates a traditional telephone network. This gives the new terminals the same usage model (creating and ending a session for each transaction) as current terminals using the traditional telephone network.

Establishing a new secured SIP session for each transaction then ending the session after the transaction is complete offers several advantages over a persistent session that is used for all transactions. Note that clients would never need to send multiple transactions in parallel; in the case of an EFTPOS pin pad terminal for example, only a single customer would be using it at a time.

A persistent transport layer connection (and therefore session) may be dropped if no traffic is sent after a long period. Thus, when the client comes to use the connection, it will find that the connection is down and would need to establish a new connection regardless. With a dynamic connection, the application can be sure that the connection exists and works when it comes to use it, as it has just established it.

Applications that use persistent connections may use keep alives or heartbeats to ensure the connection is not dropped. Using frequent keep alives, however, gives passive observers more ciphertext data for traffic analysis.

Establishing a new secured connection for each transaction ensures a different key is used for each. With a persistent connection, a rekeying interval needs to be specified (for example after n transactions, or perhaps even after every transaction). This issue is avoided altogether when using a new session/connection for each transaction. This assumes that the underlying random number generator in the TLS implementation does not generate random numbers that were recently used.

The only disadvantage of using a new session/connection for each transaction is the overhead in establishing and ending each one (see Figure 5.4). Whether establishing a single connection is time consuming will be revealed in the testbed.

Consequently, the connection between clients and the SBC are dynamic; they are established when needed, and closed when the transaction is finished. When the client wishes to send a transaction (having registered with the SBC), it would take the following steps:

1. TCP establishment
2. TLS establishment
3. SIP INVITE
4. SIP MESSAGE (to transfer the transaction data)
5. SIP BYE
6. TLS close notify
7. TCP close.

The connection does not need to be the same between SBCs, or between the SBC and the transaction processor (Figure 5.3) as it is between clients and the SBC. This is because the trust model is different; every node in the IP PSTN core network is trusted (as the SBCs control access). While the connection characteristics (more specifically the use of dynamic TCP sessions) could be the same, the SBCs could also use persistent SCTP associations to other SBCs.

In Figure 5.3, SBC1 aggregates three TCP sessions into a single SCTP association to SBC2. SBC2 could have a persistent SCTP association to the transaction concentrator. The concentrator terminates connections from multiple SBCs, ensuring the transaction processor experiences no overhead from client connects/disconnects; the processor only sees a continuous stream of transaction messages.

Trusted concentrators can also be placed before the SBC as shown in Figure 5.5 (for example at a large shopping mall). The connection between the trusted mall concentrator and the nearest SBC could also be a persistent SCTP connection. This can contribute to scalability, resulting in the SBC viewing the concentrator still as many parallel transactions (for example two shops at the mall can send a transaction at once), but without the overhead of establishing an independent connection for each. SCTP has the advantages as described in Section 3.4, such as multistreaming and multihoming.

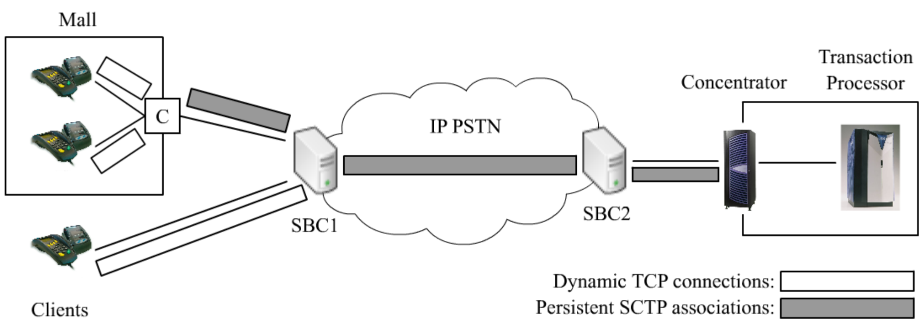


Figure 5.5: In this instance, C is a trusted concentrator, and can therefore decrypt the transaction traffic. If a concentrator on the client end is warranted, the connection between the trusted concentrator and the client's SBC would be a persistent SCTP association.

While encrypted keep alives can give observers more data for traffic analysis, SBCs would not grant IP PSTN access to potential intruders/observers.

Since SBCs and transaction processors deal with many sessions at once, the overhead in establishing and ending multiple connections may become significant. Using a persistent SCTP association avoids this overhead. SCTP's multistreaming may contribute to the performance (as each exchange between the client and SBC would in effect get its own dedicated stream) and multihoming would contribute to reliability between SBCs.

A persistent TCP connection could also be used, which would also avoid this overhead. It would have SCTP's benefits of aggregating multiple sessions into a single TCP session, although to a lesser extent, as TCP cannot replicate multiple SCTP streams in a single connection (see the Head of Line blocking discussion in Section 3.4). If dynamic TCP sessions were used between SBCs or between SBCs and the transaction processor (Figure 5.6) and were not aggregated, dedicated hardware accelerators may be required to handle the establishment and closing of multiple TLS connections at once. To replicate the multihoming function of SCTP, the responsibility and knowledge of using an alternative address would be moved up from the transport layer to the application, using the presence information provided by SIMPLE.

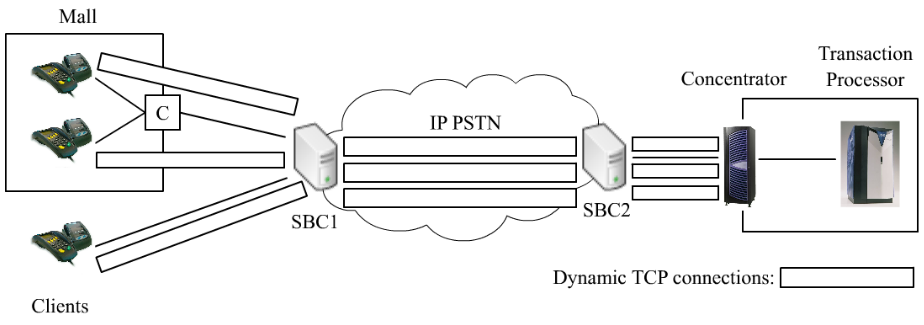


Figure 5.6: A solution using only dynamic TCP connections. As there is no aggregation before the concentrator, the infrastructure must be powerful enough to handle large numbers of TLS connections.

As stated in Section 5.2.2, end to end encryption cannot be used because SBCs need to be able to modify the packet contents. The terminals would have a tunnel to the SBC; the SBC would have a different tunnel to the destination SBC and the destination SBC would have its own tunnel to the transaction processor network. Using separate independent tunnels means if SBCs do use persistent connections, the tunnel between the client and the client's SBC can be established and closed when transactions need to be sent, while the tunnels between SBCs remain persistent. This adds to reducing connection establishment and closing overhead.

5.4.2 Open Internet Solution

While the Internet model is not the preferred solution, it is still worth analysing how capable it could be when compared to a network based on an IP PSTN.

While the Internet is ubiquitous, the infrastructure for an IP PSTN may not always be available. A solution based on the open Internet model could be arranged as in Figure 5.7. Clients connect via their ISP directly to the transaction processor/concentrator.

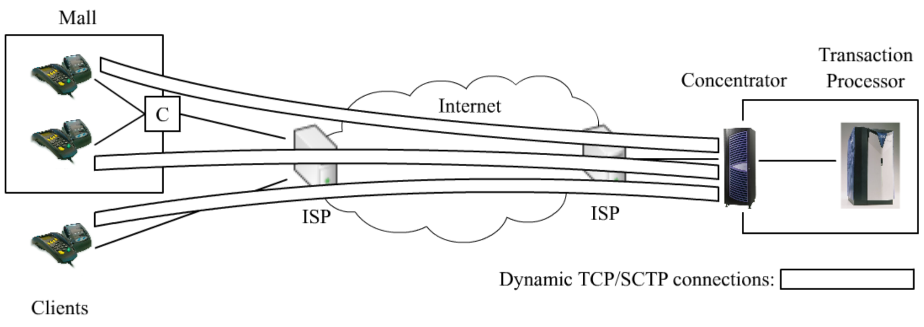


Figure 5.7: As no infrastructure is trusted on the Internet, the TLS tunnels span from the client directly to the concentrator.

Unlike the IP PSTN solution, SIP and the associated infrastructure for establishing sessions is not necessary. This is because there are no SBCs to traverse, or explicit SIP sessions needed. Take for example an internet banking transfer; the customer's computer does not need to establish a SIP session with the banking website before transferring the funds. However, this solution would still benefit from SIMPLE's MESSAGE (to frame messages when using a stream protocol such as TCP) and presence functionality (to bring network availability knowledge).

TLS would be used as the security protocol for the same reasons as used in the IP PSTN model. A full VPN solution is not required, as arbitrary network access is not necessary when the only application is transaction traffic. TLS does not require login access to the receiving server to create a tunnel. As noted in the IP PSTN solution, the creation of SSH tunnels require login access to the receiving server. This opens the receiving server to unnecessary risk. This risk is greater in the Internet model, as any client connected to the Internet can attempt to access the transaction processor.

Authentication is also two way, via certificates, to ensure that only permitted clients can create a connection to the transaction processor. In an e-commerce situation, only the server is authenticated, as there is no universal, efficient way to distribute client side certificates. In this situation, however, the transaction terminals will have been issued with a certificate at the time of manufacturer by the trusted device vendor (see Section 4.3.5).

Figure 5.7 shows the encrypted tunnels span from the client device directly through to the transaction processor. Like the IP PSTN solution, a concentrator may be physically placed between the network and the actual transaction processor. The tunnel is not terminated in any way before it arrives at the transaction processor, as no infrastructure connected to the Internet is trusted (unlike the SBCs in the IP PSTN model).

Again, like the IP PSTN solution, the connection between the terminals and the concentrator could be either TCP or SCTP. Both are reliable transport protocols with compatible flow and congestion control mechanisms. Transactions would not be aggregated while being routed through the Internet, as no infrastructure would be trusted to decrypt the transactions and combine them. This is not a major problem, however, because the devices connect directly to the concentrator; each device can only send a single message at a time (for example only one customer can be entering their PIN into a keypad at a time).

The concentrator would then aggregate these connections onto a single SCTP association, directed to the transaction processor. The connection between the concentrator and the transaction processor could also be a TCP connection as in the IP PSTN solution. This would require SIMPLE's MESSAGE application layer framing in order for the transaction server to determine where each transaction started and ended. The concentrator would require hardware that could handle multiple TLS tunnels at once to ensure adequate performance.

For high transaction rate sites (for example a large shopping mall), a concentrator can also be placed in front of the transaction server's concentrator (Figure 5.8). This is identical to the site concentrator in Figure 5.5. If the transaction rate is high enough to warrant a site concentrator, then it would use a persistent SCTP connection to the transaction processor's concentrator.

Once more like the IP PSTN solution, the connection between the devices and the destination (the concentrator in this case) would be dynamic. A concentrator would be unable to support a persistent TLS secured tunnel for every possible transaction device on the network. The justifications for using a dynamic connection in the IP PSTN solution also apply in this solution. Idle connections may be dropped after a certain period. Heartbeats or keep alives may be used to keep connections alive, but these offer more ciphertext data for traffic analysis. On the open Internet, where nothing is trusted, this becomes a significant risk.

When a client wishes to send a transaction, it would take the following steps:

1. TCP establishment

2. TLS establishment

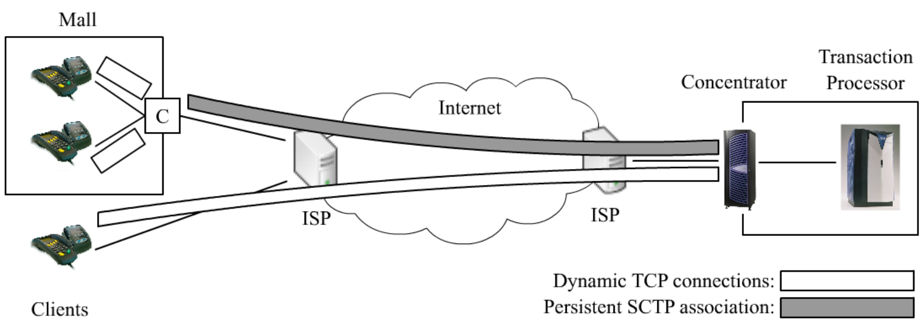


Figure 5.8: In this instance, C is a trusted concentrator, so has the ability to decrypt transaction traffic. It aggregates these onto a single SCTP association, reducing the load on the transaction processor's concentrator.

3. SIP MESSAGE (to transfer the transaction data)
4. TLS close notify
5. TCP close

Unlike the IP PSTN model, where only trusted clients could access the core network and other clients, the Internet has no access restrictions. This opens the Internet based solution to all forms of DoS attacks, including Distributed DoS (DDoS) and Distributed Reflection DoS attacks (DRDoS). DoS attacks are identified as either exploiting a network problem or an individual problem in [90]. Network problems are based on abusing flaws in Internet protocols and infrastructure. Individual problems focus on taking advantage of poorly configured systems and harnessing them for use as attack zombies.

Improving the infrastructure can mitigate network problems, for example load balancing and traffic queuing [73]. Protocols can be altered as well, for example using SYN cookies to avoid SYN flooding (as described in Section 3.3.3). Cryptographic puzzles can be used to impose a computational cost to clients wanting to establish a TLS connection with a server; a method to reduce the computational load on the server when performing a TLS handshake is described in [35]. As the TLS receive rate increases, the complexity of the puzzle increases, giving the server a constrained means to limit the send rate of clients.

Honeypot or Honeynet diversion networks can be deployed within the DMZ part of the network as described in [90]. IDSs can be installed on the network, triggering alerts when unusual network activity occurs.

Improvements in network infrastructure and protocols alone, however, cannot protect from DoS attacks completely; individual computers and users must also take measures to protect against DoS attacks. Antivirus and anti-malware software should be used to ensure computers and equipment do not become infected and are used as launch pads for DoS attacks. Firewalls should be configured for ingress/egress filtering (for example ensuring that traffic leaving the network does not have an external source address and incoming traffic does not have an internal source address). Operating systems and software must also be kept up to date with the latest patches and bug fixes.

Ultimately, the problem stems from the inability to identify and separate malicious traffic from legitimate traffic. Previous work found the best solution was to detect the attack as early as possible, trace the IP address of the source then notify the administrators of the zombie machines [90]. However, they concluded there was no 100 percent effective method for defending against DoS attacks.

5.4.3 Conclusion

A transaction network can be based on an IP PSTN or the open Internet model. The two architectures described meet all of the qualitative requirements set out in Section 2.6. Whether these architectures meet the quantitative requirements (such as scalability, throughput and response time) can only be determined after a proof of concept testbed has been created and benchmarked. Table 5.2 summarises the mapping of these solutions to the requirements set out in Section 2.6. It is interesting to note that despite the differences in the network models, the solution architectures are very similar. The main point of difference is in the QoS guarantees and security. Only the IP PSTN can provide QoS guarantees; the Internet is a best effort service. In Table 5.2, the Send Rate Limited row alludes to DoS attacks, as DoS attacks rely on overwhelming the server's bandwidth or system resources. As the Internet is an uncontrolled network, the risk of DoS attacks can never be removed, only reduced.

Requirement	IP PSTN model (client to SBC)	IP PSTN model (between trusted infrastructure)	Open Internet Model
Transport Protocol	TCP or SCTP	SCTP	TCP or SCTP
Security Protocol	TLS	TLS	TLS
Connection Type	Dynamic	Persistent	Dynamic
Message Oriented	Yes*	Yes	Yes*
Transaction Aggregation	Yes**	Yes	Yes**
Reliable Transport	Yes	Yes	Yes
Network Reliability	Yes*	Yes (via multihoming)	Yes*
Link Failure Alarms	Yes*	Yes (via built in keep alives)	Yes*
Flow and Congestion Control	Yes	Yes	Yes
Vulnerable to SYN flood DoS?	No (when used with SYN cookies)	No	No (when used with SYN cookies)
Send Rate Limited?	Yes (by SBC)	Yes (by SBC)	No

Table 5.2: Mapping of the IP PSTN and Internet solution architectures to the requirements in Section 2.6. **not needed when a client can only send a single transaction. *SIMPLE required for TCP.

Chapter 6

Architectural Implementation

This section describes the architectural validation via a proof of concept network and justifies design decisions in the implementation. The network aims to demonstrate the viability of the two architectures: IP PSTN and Internet, described in the previous section.

Validating the architectures required the use of a live test bed network instead of a simulation. This is because network simulations cannot accurately model specific application level protocol exchanges. A network simulation also does not demonstrate the chosen messaging and security protocols in operation.

The client, concentrator and trasaction processor were all written in Java. The JAIN-SIP library[1] was used. JAIN-SIP is an open source implementation of RFC 3261. The client was initially written in C, using the Sofia-SIP library[2], also an open source SIP library. A native C/C++ implementation is generally faster than runtime compiled implementations (for example Java and C#). However, Java was ultimately chosen, as the JAIN-SIP library was easier to use. Performance differences between the languages are also insignificant when compared to the impact imposed by encryption, digital signatures and network latency.

[1]`https://jain-sip.dev.java.net`

[2]`http://sofia-sip.sourceforge.net`

6.1 The Client

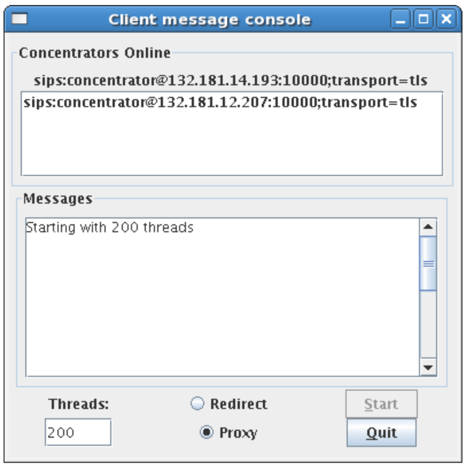
The client software initiated sessions to send transactions. The most accurate approach to demonstrate the network would be to utilise a large number of physical clients, each running a single instance of the client software. This was not feasible, however, so the client software was multithreaded, using each thread to represent a single instance of the client. A rudimentary user interface was created to separate transaction progress from error messages (Figure 6.1).

The 'Start' button initialised and ran a number of threads, specified by the number entered in the 'Threads' field. From herein, 'client' refers to a particular instance of a thread. Thus, clicking 'Start' for 200 threads approximated 200 clients. Each 'client' or thread ran through the exchange in Figure 6.3 or Figure 6.4 (depending on the architecture selected) in parallel with the other threads (that is, the threads were not run sequentially one after the other). Message payloads were 300 bytes long (as stated in [14]). Timeouts were set to 5000ms; if the client did not receive a response to any request within 5000ms, it assumed the server was down and tried another (in the Internet model) or retransmitted (in the IP PSTN model). The transaction timeout was set to 5000ms as it was considered an acceptable time to wait for a transaction to go through; transaction terminal users can take this long to enter their PIN. While lower timeouts would improve performance in the event of transaction processor failure, lower times could also unnecessarily congest the network; the client could potentially consider the transaction processor nonfunctional, when in reality it was simply taking longer than usual to complete the transaction.

The network implemented here differs slightly from the architectures described in Section 5.4. Section 5.4 described the use of SIMPLE infrastructure to provide service awareness. In this network, a SIP Registrar provides service awareness. This is because SIMPLE is designed primarily for IM systems, which this system is not, although it has some IM-like attributes.

Section 3.6.4 described the SIMPLE infrastructure; users (or servers in the case of this network) would send periodic REGISTER messages indicating their status (for example online or offline). The presence server, upon receipt of these messages, would send NOTIFY messages to each user (clients, or transaction terminals in the case of this network) which had subscribed to receive notifications.

IM systems, for which SIMPLE was designed, benefit from knowing at all times, which users are online. For example, a user who has signed onto their IM application will benefit from knowing which friends on their contact list are online, regardless of whether he or she wishes to send them an IM.



sips:concentrator@132.181.14.193:10000;transport=tls
sips:concentrator@132.181.12.207:10000;transport=tls

Starting with 200 threads

Threads: 200

Redirect [✓ Proxy]

Figure 6.1: The server at the top of the window (in this case 132.181.14.193) is the server currently being used by the client. The list underneath contains back up servers. The messages window shows transaction progress; errors are printed to the command line. The redirect and proxy buttons switch the client between the IP PSTN model (Proxy) and the Internet model (Redirect).

Clients in this network, on the other hand, only need to know the status of the receiving server when sending a transaction. SIP Registrars are sufficient to provide this service, making periodic alerts (in the form of NOTIFY messages) an unnecessary use of bandwidth and processor load.

6.2 The Concentrator

The concentrator aggregated sessions onto a single session. The aim was to reduce the computational load on the transaction processor associated with multiple connects/disconnects. The concentrator appeared as a transaction processor which clients connected to. As a result it sent periodic REGISTER messages to the SIP registrar like a transaction processor. Concentrators, however, did no actual processing and simply proxied transactions between transaction processors and clients. Like the other components, the concentrator was multithreaded to enable the parallel processing of requests.

When the concentrator first came online, it established a session with the transaction processor it was associated with, regardless of whether transactions were arriving. When a client connected to the concentrator, a session was established between each client and the concentrator. When the clients sent their transactions in MESSAGE payloads, the concentrator forwarded only these MESSAGE requests to the transaction processor. Responses were also forwarded back to the client. When the transaction was complete, the clients ended their session with the concentrator, however the session between the transaction process and the client remained.

This was expected to reduce the load on the transaction processor, as only a stream of MESSAGE requests were received; the transaction processor was not exposed to the overhead of multiple client connects and disconnects. The concentrator and transaction processor thus had a persistent session as illustrated in Figure 5.4.

Unlike the solutions described in Section 5.4, where the persistent connections associated with concentrators used SCTP, the concentrators here used TCP. This was because the JAIN-SIP library did not support SCTP on the operating system used (Fedora Core 6). The functionality was the same, however, as the MESSAGE requests provide application level message framing for TCP, which is provided at the transport layer in SCTP.

6.3 The Transaction Processor

The transaction processor sent REGISTER messages every 10 seconds to the SIP server, each with an expiry of 15 seconds. For a network of the size demonstrated, these times were low enough to get reasonably fast notification of transaction processor status, without overwhelming the network. More frequent REGISTER messages with a shorter expiry time offered more accurate transaction processor status information at the cost of higher network bandwidth and system load. Less frequent REGISTER messages with a longer expiry time would have resulted in a lower system and network bandwidth load, but a higher chance of a client attempting to contact an offline transaction processor. For example, if expiry timers were set to 1 minute, the SIP registrar would consider the transaction processor available for at least a minute, even if the transaction processor went offline immediately after sending the REGISTER message.

The transaction processor received transactions and returned replies. The transaction processor was also multithreaded, enabling the processing of multiple clients simultaneously. In the testbed network, clients could connect directly to transaction processors, rather than through a concentrator as transaction processors and concentrators can be considered a single logical unit (as noted in Section 5.4). They have been separated here in order to analyse the benefit of aggregating sessions.

6.4 The SIP Server

The SIP server used was OpenSER 1.2.0-tls.[3] OpenSER is an open source variant of Sip Express Router (SER).[4] OpenSER has a focus on new features, frequent minor releases and was created in response to SER's slow development and long release cycles. OpenSER also has integrated TLS support, while SER's TLS support is in a separate patch.

The SIP server operated as a SIP registrar. In the Internet model, it also operated as a redirect server, while in the IP PSTN model, it operated as a proxy server.

To be faithful to the models proposed, a true SBC should have been used. However, no SBC appliances or appropriate SBC software was available. OpenSipStack's OpenSBC was considered[5] however it did not support SIPS or TLS connections. As security was a requirement, OpenSBC was not used.

6.5 IP PSTN Network

In this architecture, the client sent and received all messages to and from the SIP server. The client communicated with the SIP server through a secure TLS session. The SIP server also maintained an independent TLS session with each transaction processor it communicated with. The architecture explained here is an implementation of Figure 5.6. This was because the SIP server software did not support SCTP at the time of writing. The JAIN-SIP library also did not support SCTP on the operating system used (Fedora Core 6).

Application Protocol
SIP
TLS-TCP
IP

Figure 6.2: The protocol stack used in the test bed network.

The solution stack used in this model is shown in Figure 6.2. The Application Protocol layer represented the purpose of the network, for example EFTPOS has its own protocol, AS2805. In a deployment, AS2805 would be the application protocol, that is, AS2805 could be carried as the payload in a MESSAGE request. In this test bed network, however, 300 bytes of text was

[3]`http://www.openser.org`
[4]`http://www.iptel.org`
[5]`http://www.opensipstack.org`

used as the MESSAGE payload, as the goal of this project was to define and demonstrate a suitable protocol stack for short generic transaction networks as opposed to implementing EFTPOS over IP.

The client created an INVITE for a transaction processor and sent it to the SIP server (message 3 in Figure 6.3). The SIP server performed a look up, and relayed the INVITE (4) to the transaction processor. The transaction processor responded with a 200 OK (7), which was relayed back to the client, which completed the handshake with an ACK request (13 and 14).

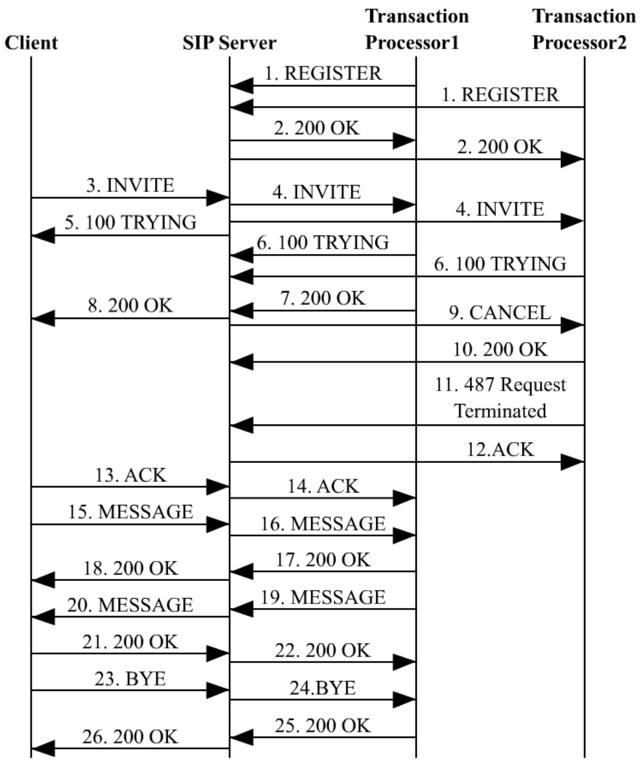


Figure 6.3: A typical exchange in the proxied IP PSTN model. This is an implementation of the solution illustrated in Figure 5.6.

The client then sent a MESSAGE request with the transaction payload (15). The SIP server relayed this to the transaction processor, which responded with a 200 OK (17 and 18), to acknowledge it had received the MESSAGE request. The transaction processor then checked whether the request was successful or not, and sent its reply in the payload of a MESSAGE request (19). The SIP server relayed this back to the client (20), which also responded with a 200 OK (21 and 22). Regardless of whether the transaction was successful or not (for example ‘insufficient funds’), the client ended the session by sending a BYE request (23 and 24) for the transaction processor to the SIP server. When the transaction processor received this, it responded with a 200 OK (25 and 26) and the session was terminated.

The SIP server forked the client’s INVITE to each transaction processor that had registered (4). The SIP server returned the first 200 OK it received to the client (7). Thus the first transaction processor that replied with a 200 OK was given the call. The client then responded with an ACK request which was sent to the SIP server (13) and proxied to the transaction processor (14), to establish the session. This provided a degree of load balancing as servers which respond to a 200 OK first are less likely to be heavily loaded, and it is these servers which accept the call.

When a transaction processor received an INVITE request, it replied with a 100 TRYING message (6), then a 200 OK message (7). A 200 OK message is classed as a final response, while 100 TRYING is a provisional response. RFC 3261 stipulates that proxies must return all 200 OK and other final responses for a forked INVITE to the client, while provisional responses (such as 100 TRYING) do not need to be returned.

If the SIP server has not received a final response to its forked INVITE, it can use a CANCEL request to cancel establishment of the session (9). If it has received a final response, however, it must be passed back to the client, which must use a BYE request to terminate the session; CANCEL requests can only be used to cancel sessions for which no final response has been received. Consequently, when a transaction processor responded with a 200 OK, the SIP server relayed the response back to the client and sent CANCEL requests to all the other transaction processors it forked INVITE requests to, in order to terminate the pending INVITE requests.

Transaction processors could have returned 200 OK messages instead of 100 TRYING messages in response to an INVITE. However, if the SIP server had forked to many servers which were online, it would have relayed many 200 OK messages back to the client. The client would then need to ACK one 200 OK response, and send a BYE request to every other 200 OK response it received. If there were many servers online, this may have placed a heavy load on the client. Thus 100 TRYING messages gave the SIP server an opportunity to cancel requests, thereby relieving the client from having to send multiple BYE requests for each online server which received a forked INVITE. TRYING and other provisional messages are used to indicate that an operation is in progress; if there were no provisional messages available, clients would only know the server on the other end was online when the operation was completed. Operations that require a slightly longer time maybe interpreted by the client as a timeout.

The SIP server forked the INVITE requests in parallel; it relayed the INVITE message to every registered transaction processor simultaneously. The first transaction processor to respond with a 200 OK accepted the call, resulting in lower response times. The disadvantage is that multiple transaction processors may respond with a 200 OK before the SIP server has a chance to CANCEL the pending INVITE request. This placed responsibility on the client to terminate the unnecessary sessions with BYE requests, which resulted in extra load for the client.

The alternative was serial or sequential forking, where the SIP server sent the INVITE to the transaction processors one at a time. If request failed, timed out or the transaction processor went offline, the SIP server sent the INVITE to the next registered server in the list. The advantage of this is that the client would only ever receive one 200 OK in response to sending a single INVITE request. The disadvantage is that transaction processors at the top of the list may have gone offline before their registration had expired. The SIP server would then contact the offline servers and wait for them to timeout before sending the INVITE to the next one. This would have resulted in a longer response time.

Forking brings fault tolerance during the session initiation stage; if a transaction processor fails before its registration expires, the SIP server will have forked other transaction processors (in the case of parallel forking) or will do so when the current forked INVITE times out (in the case of sequential forking).

However, transaction processors can also fail during a transaction. Eventually the session will time out, signalling the failure of the transaction processor to the client. As long as the transaction did not complete or commit, the client will resend the INVITE to the SIP server, which will relay it to the next available transaction processor. If there were no available transaction processors, the SIP server would redirect the client to another SIP server/SBC (described in Section 6.6), which may have operational transaction processors registered with it. Failure of all SIP servers/SBCs would constitute a major network disaster; the SIP servers/SBCs are a crucial network infrastructure in this model. The SIP servers/SBCs represent a central point of failure and are a drawback of this architecture, as discussed in Section 5.2.2.

6.6 Open Internet

In this architecture, the client sent and received messages directly to and from the transaction processor (Figure 6.4). Clients may be configured with the addresses of transaction processors, or SIP redirect servers. The architecture explained here is an implementation of the solution illustrated in Figure 5.7. TCP was used, for the same reasons as in the private IP PSTN model; SCTP was not supported in the SIP server, nor did the JAIN-SIP library provide SCTP connections on the operating system (Fedora Core 6). The solution stack was the same as the IP PSTN model, in Figure 6.2.

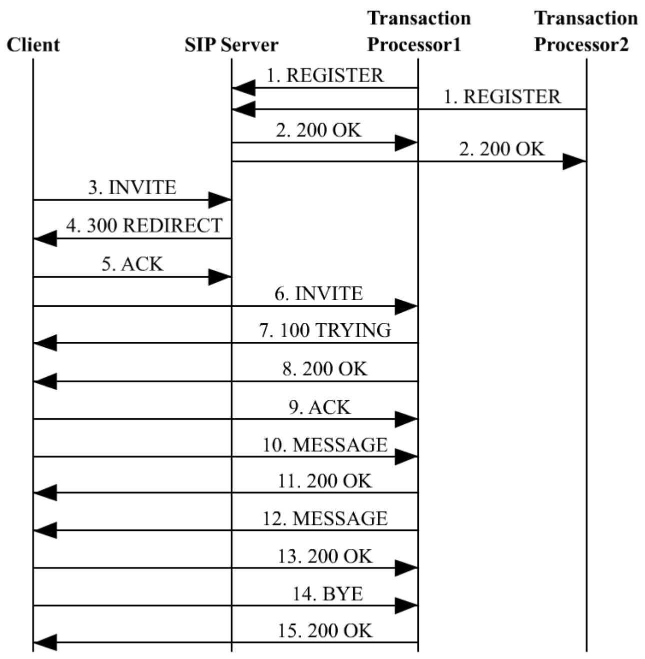


Figure 6.4: A typical exchange in the Internet model, which uses redirects.

If clients had no knowledge of the addresses of transaction processors, they would contact the SIP redirect server first. The SIP server in this model operated as a registrar and location server. Clients would need to contact the SIP server in order to get a list of available transaction servers. SIP's method is to use an INVITE request, and as illustrated in Figure 6.4, a 300 REDIRECT message would be sent in response by the SIP server. The awkward alternative would be to use a MESSAGE request, and have the SIP server store the list in a MESSAGE payload in response.

REDIRECT responses contain contact bindings for a particular user, that is, addresses of ‘users’ that have registered as transaction processors. The REDIRECT response can also include the address of other SIP servers. The client stored this list internally, and sequentially forked INVITE requests until it located an operational server. The client differentiated between a SIP server and a transaction processor by the response it received in addition to the URI; if it received a 200 OK in response to an INVITE request, it had contacted a transaction processor.

The client could dispense with needing to create a session altogether and send MESSAGES immediately as required. MESSAGE requests could be sent, without first establishing a session with an INVITE request, even with no information on available transaction servers. If this method was used, Message 3 in Figure 6.4 would be a MESSAGE request instead of an INVITE request. A 300 REDIRECT message with contact bindings would still be sent in response. The client would then repeat, sending the MESSAGE request to a server in the list. However, using MESSAGE requests (which contain the sensitive transaction details as a payload) to probe for online servers creates unnecessary risk.

Regardless of requiring strict two way TLS authentication (which ensures the client is communicating with a trusted entity), revealing transaction payload details unnecessarily to any network entity other than the proper receiving transaction processor opens the possibility for fraud. For example while only trusted infrastructure will have the required certificates to enable successful TLS negotiation, the proxy software could be modified to record all traffic sent and received. This would result in giving a trusted entity, which has nothing to do with the processing of transactions, knowledge of the transaction data.

The subsequent messages exchanged are identical to the IP PSTN model, except messages are sent directly to the transaction processor, instead of being proxied through the SIP server.

If a transaction processor fails during the INVITE stage, the client will use the next server in its list (TransactionProcessor2). If all the servers in the list have been exhausted, the client will contact the SIP server again (as other transaction processors may have come online since the last contact).

If transaction processors fail mid transaction, as long as the transaction did not complete or commit, the client will start a new session with the next server on the list. If the client tries all servers on the list without success, it will contact the SIP server again to get an updated list. In this model, the SIP servers are analogous to DNS servers, which provide a look up location service. If the SIP redirect servers failed, as long as the addresses of the transaction processors were known, the system would still function as usual. This is much like DNS, where even if the DNS service failed, web browsing would still work if the IP addresses were known.

Chapter 7

Architectural Validation and Evaluation

7.1 Network Setup

The Client was a Pentium III 500MHz with 512MB RAM. The SIP server was a Pentium 4 2.4GHz also with 512MB RAM. The Concentrator and Transaction Processor were both Athlon 800MHz machines with 768MB RAM. The machines were connected to a 10Mbps hub/Ethernet repeater to create an isolated LAN.

All of the computers ran Fedora Core 6 without running any other processes which could have affected processor or network utilisation. Java 5.0 (1.5.0) was used to run the client, concentrator and server software.

7.2 Experimental Design

As described in Section 1.1, this project's main aim was to define a generic IP based transaction architecture and demonstrate a proof of concept system. Experiments were run to gauge the performance of each architecture. AES and 3DES ciphers were used to encrypt, as 3DES is commonly implemented in hand held payment terminals, while AES is commonly adopted as 3DES's successor. To avoid the effects of memory caching, disk paging and Java's JIT dynamic compilation optimisations, the first five readings of each experiment were not recorded. Each experiment was then repeated 10 times with the times recorded. Several experiments were conducted, and methodology for each are detailed below. Results were analysed using ANOVA at the 95% confidence interval.

7.2.1 Single Transaction Measurements

This experiment recorded the total and CPU time as well memory usage on the client when sending a single transaction to a transaction processor. This experiment was performed with both the IP PSTN and Internet architectures with various security levels. As only a single transaction was transferred, concentrators were irrelevant to this test. The total time was recorded from the creation of the client thread to the acknowledgement of the client's BYE, terminating the session. Thus, this included network latency and transaction processor time. The CPU time was recorded as the sum of the user and system time. This provided details on the loading of the client.

Memory usage was measured using Java's `jmap` memory map utility. The heap usage was recorded as the sum of all heapstores used (Eden, From, To, PS Old Generation and PS Perm Generation spaces). While the memory footprint of a Java implementation will always be larger than a native implementation, this test gave details on the relative effect of enabling encryption. That is, if encryption had a larger Java heap footprint than the unsecured exchanges, under the same circumstances (for example no hardware acceleration or dedicated hardware support), encryption would also have a larger footprint in a native implementation. Thus, the memory tests did not aim to give an indication of how much memory the implementation required on an embedded device as any deployment would use a native implementation.

This experiment also provided insight into how transaction time was proportioned, that is, how much time was spent establishing the session, performing the actual transaction and ending the session. This experiment was performed with no security (to provide a base case), AES+SHA TLS encryption and 3DES+SHA TLS encryption to ascertain the extent of processing overhead imposed by encryption and message authentication.

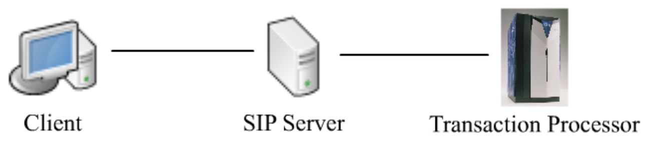


Figure 7.1: Transaction path for the IP PSTN architecture. The SIP server represents the SBC.

The 500MHz client machine has similar performance to high end embedded processors. Intel XScale processors running at 1GHz achieve approximately 1250 Dhrystone 2.1 MIPS [40], while a 500MHz Pentium III achieves between

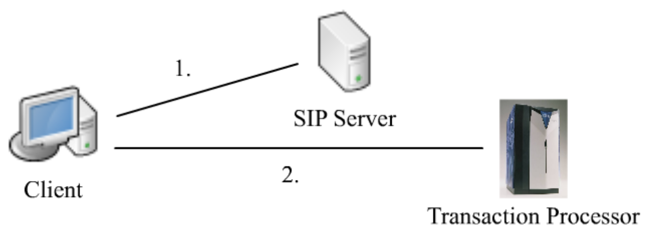


Figure 7.2: Transaction path for the Internet architecture. The client populates the server list by contacting the SIP server before it can connect to a transaction processor.

1200-1700 MIPS.[1] The transaction path is illustrated in Figures 7.1 and 7.2 for the IP PSTN and Internet architectures respectively. Figure 7.1 represents an implementation of Figure 5.6 without the concentrator; clients connect directly to the SBC (SIP server in this case) and the SBC proxies the connection to the transaction processor. Figure 7.2 represents Figure 5.7 also without the concentrator; clients first contact the SIP server (not shown in Figure 5.7) before connecting directly to the transaction processor. The factors in this test were consequently three security schemes x two architectures.

7.2.2 Multiple Transaction Measurements

This experiment also measured the total and CPU time and memory usage on the client, but this test transferred 50 threaded transactions. This approximated 50 transaction terminals on the client machine. This test gave a rough indication of how long slower embedded processors would take with encryption and message authentication. As a rough guide, a Motorola MC68040 processor is approximately 1/50th the performance of a Pentium III 500MHz.[2]

This experiment was also performed using both the IP PSTN and Internet architectures, as well as the same three security levels in the previous test: no security, AES and 3DES. This experiment used the same transaction path as the single transaction experiments; Figures 7.1 and 7.2 show the transaction path for the IP PSTN and Internet architectures respectively. The only difference was the number of threads run on the client.

Note that only one transaction processor was operating. This ensured that the load balancing provided by the SIP Proxy/SBC had no effect here. The

[1]`http://www23.tomshardware.com/cpu_2004.html`

[2]`http://www.tahi.org/lcna/docs/cpu-performance/processor.html`, `http://www.skepticfiles.org/cowtext/comput~1/486vs040.htm`

time was recorded from the creation of the first thread to the termination of the last session, while CPU time is again the sum of user and system time on the client. The factors in this test were three security schemes x two architectures.

7.2.3 Impact of Concentrators

While the first two tests only measured total and CPU time as well as memory usage on the client, this test also measured CPU time and memory usage on the transaction processor. This test revealed whether a concentrator had any benefit for transaction processors by hiding client connects/disconnects (as described in Section 6.2). The client transferred 50 transactions as in the second experiment. The architecture was not important here as the concentrator does not care whether the transaction comes directly from a client terminal or a SIP Proxy/SBC. Thus the Internet architecture was used in this test to remove any effects the SIP Proxy may have had. When unconcentrated, the transaction path is that depicted in Figure 7.2. Figure 7.3 shows the transaction path with a concentrator. This set up represented an implementation of Figure 5.7. The factors in this test were three security schemes x Concentrated/Unconcentrated.

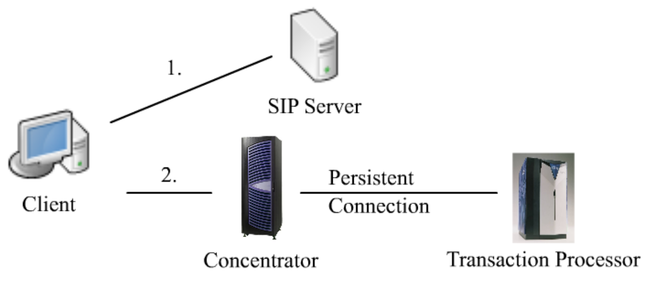


Figure 7.3: Transaction path with concentrated sessions. The concentrator maintains a persistent session with the transaction processor.

7.2.4 Recovery Time

This test demonstrated the feasibility of the recovery mechanism and measured the time required to recover from a transaction server failure for the two architectures. The CPU time on the client was also measured. Unlike the other experiments where only one transaction processor was operating, this test had two transaction processors operating. AES encryption was used for both tests as security was not assumed to be a factor in recovery time. The client sent a single transaction and the first transaction processor was made to fail after receiving the MESSAGE request from the client. The opposing method would be to have one server fail randomly, however this could have reduced the comparability between the two models. The transaction path is shown in Figures 7.4 and 7.5 for the IP PSTN and Internet models respectively.

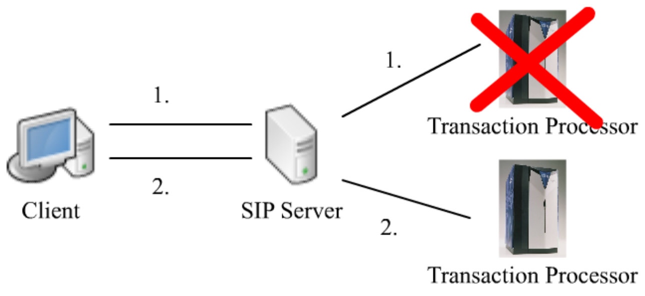


Figure 7.4: Transaction path with transaction processor failure in the IP PSTN architecture. After the client times out, it starts a new session. The SIP server then proxies the transaction to the working transaction processor.

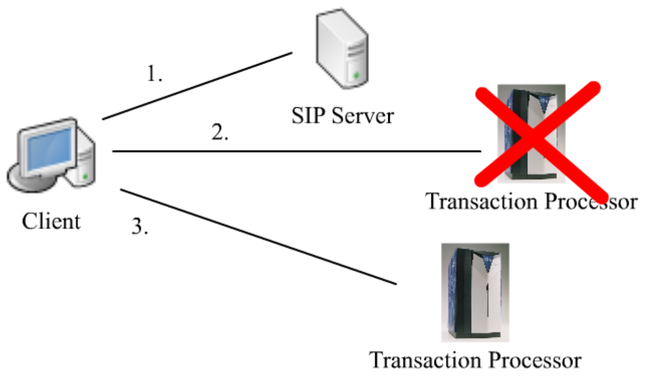


Figure 7.5: Transaction path with transaction processor failure in the Internet architecture. After the client times out, it starts a new session with the next transaction processor in the list.

While a transaction timeout of 5000ms was used for all previous experiments, three timeout values were used here. Network architecture was also a factor in this test. Therefore, the factors in this test were three timeout values x two architectures. Testing with 50 threads would have given information on how the architecture handled a relatively larger number of transactions, unfortunately not all threads timed out properly.

Chapter 8

Results and Discussion

8.1 Single Transaction

In the first experiment, the client was configured to send one transaction using both the IP PSTN and Internet architecture. Figure 8.1 shows the mean phase and total times.

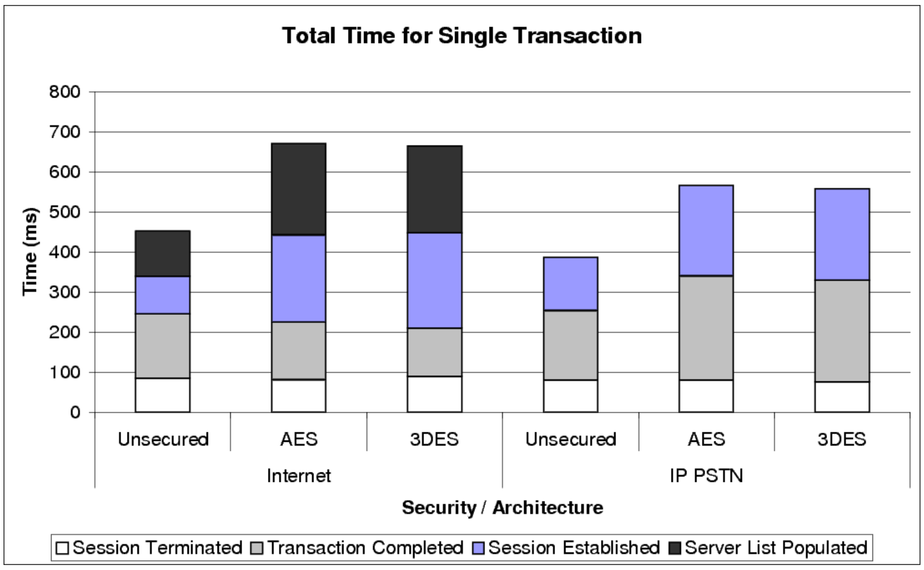


Figure 8.1: Mean times for a single transaction.

For a single transaction, the Internet architecture's total time was slower. This was due to the additional phase of populating the server list. This step was not required in the IP PSTN model. Enabling encryption resulted in slower

Security	Server List Populated	Session Established	Transaction Completed	Session Terminated	Total Time	S.D.
Unsecured	112.9	94.1	160.5	84.6	452.1	17.9
AES	227.8	217.6	144.2	80.6	670.2	53.8
3DES	217.6	237.7	120.8	88.9	665.0	54.8

Table 8.1: Mean total times for a single transaction using the Internet model. All figures are in ms.

Security	Session Established	Transaction Completed	Session Terminated	Total Time	S.D.
Unsecured	132.6	173.3	80.2	386.1	54.9
AES	225.9	260.2	79.8	565.9	46.7
3DES	228.4	251.2	75.3	557.8	54.1

Table 8.2: Mean total times for a single transaction using the IP PSTN model. All figures are in ms.

times for both architectures.

The differences between the architectures were statistically significant ($F_{1,54} = 53.70, p < 0.01$) as were the differences between the encryption schemes ($F_{2,54} = 106.84, p < 0.01$). The mean values for each phase are in Table 8.1 and 8.2 for the Internet and IP PSTN architectures respectively. There was no statistically significant interaction between architecture and the encryption scheme.

Figure 8.2 shows the mean CPU times used. The CPU times mirror the total times; the Internet architecture required more CPU time than the IP PSTN architecture due to the additional phase of discovering available transaction processors. Enabling encryption also resulted in more CPU time used for both architectures.

Again the differences between the architectures for CPU time used were

Security	Internet	S.D.	IP PSTN	S.D.
Unsecured	260.4	19.0	224.0	30.8
AES	374.8	20.1	343.2	19.0
3DES	384.4	25.7	337.7	22.1

Table 8.3: Mean client CPU times for a single transaction. All figures are in ms.

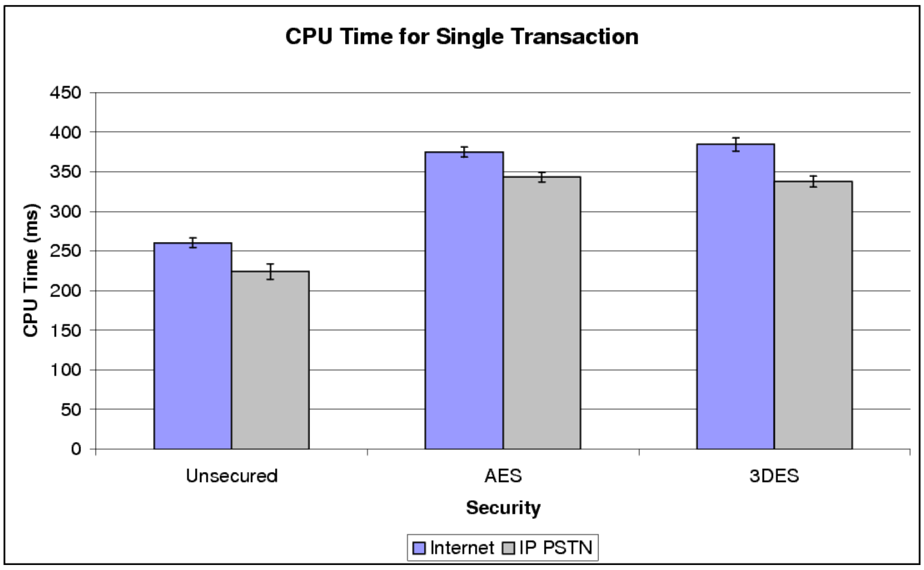


Figure 8.2: Mean client CPU times for a single transaction.

statistically significant ($F_{1,54} = 40.77, p < 0.01$) as were the differences between the encryption schemes ($F_{2,54} = 172.12, p < 0.01$). There was no statistically significant interaction between architecture and the encryption scheme in terms of CPU time. The mean values are in Table 8.3.

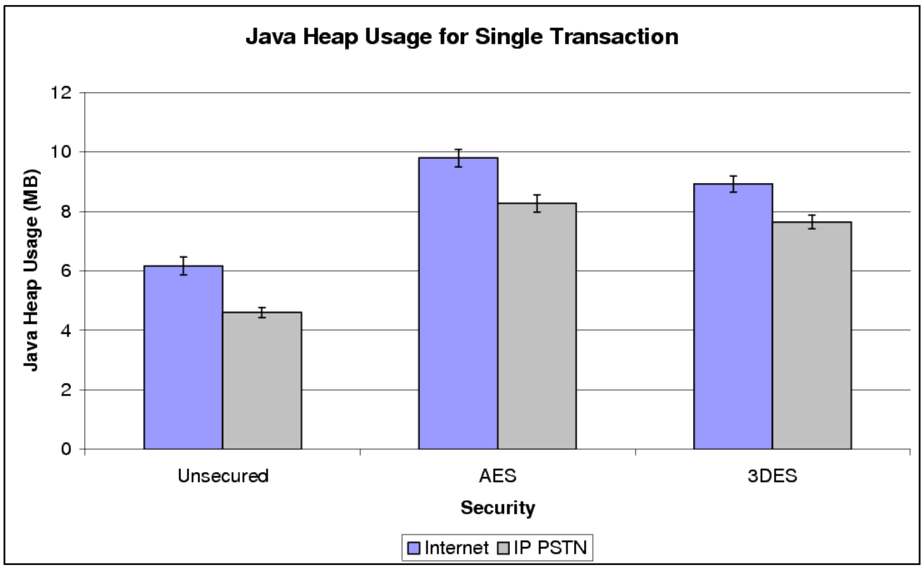


Figure 8.3: Mean client heap usage for a single transaction.

Memory/Java Heap usage followed the same pattern as the CPU time used (Figure 8.3). The Internet architecture required more heap space under all security schemes. This is also assumed to be a consequence of the additional phase. Enabling encryption increased memory usage, with AES requiring the most under both architectures. Despite Java's periodic garbage collection, the variance in heap space used was not as large as expected.

Security	Internet	S.D.	IP PSTN	S.D.
Unsecured	6.16	0.94	4.60	0.53
AES	9.79	0.93	8.27	0.92
3DES	8.92	0.83	7.64	0.72

Table 8.4: Mean client heap usage for a single transaction. All figures are in MB.

Like the other tests, the differences here between the architectures were statistically significant ($F_{1,54} = 46.70, p < 0.01$) as were the differences between the encryption schemes ($F_{2,54} = 109.14, p < 0.01$). There was no statistically significant interaction between architecture and the encryption scheme. The mean values are in Table 8.4.

8.2 Multiple Transactions

In this experiment, the client was configured to send 50 simultaneous transactions using both the IP PSTN and Internet architecture. Figure 8.4 shows the mean total times.

While the differences between the two architectures were minimal with no security, they were notable when encryption was enabled. The IP PSTN architecture was clearly slower than the Internet architecture when encryption was used. Both architectures required more time when encryption was enabled.

The differences between the architectures were statistically significant ($F_{1,54} = 171.47, p < 0.01$) as were the differences between the encryption schemes ($F_{2,54} = 472.10, p < 0.01$). There was also a statistically significant interaction between the architecture and encryption ($F_{2,54} = 43.78, p < 0.01$). This interaction is clear in Figure 8.4; while the IP PSTN architecture was slower than the Internet architecture, the Internet architecture suffered a larger performance penalty when switching from AES to 3DES than the IP PSTN architecture. The mean values for the total times are in Table 8.5.

Figure 8.5 shows the mean CPU times when 50 transactions were sent. Like

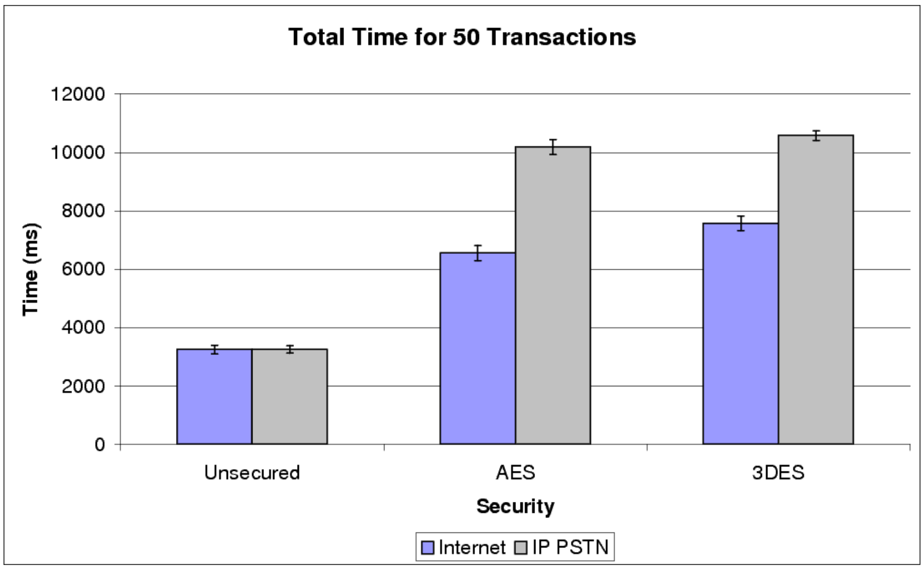


Figure 8.4: Mean times for 50 transactions.

Security	Internet	S.D.	IP PSTN	S.D.
Unsecured	3244.1	460.8	3250.1	409.5
AES	6557.0	818.8	10187.7	810.9
3DES	7572.5	782.2	10577.6	502.7

Table 8.5: Mean total times for 50 simultaneous transactions. All figures are in ms.

the single transaction experiment, the CPU times mirrored the total times. The IP PSTN model was slower than the Internet architecture when encryption was enabled. Encryption required more CPU time in both architectures.

The differences in CPU time between the architectures were statistically significant ($F_{1,54} = 32.93, p < 0.01$) as were the differences between the encryption schemes ($F_{2,54} = 302.05, p < 0.01$). The interaction between the architecture and encryption was also statistically significant in terms of CPU time ($F_{2,54} = 8.67, p < 0.01$). This effect is the same as the interaction in the total times; the Internet architecture had a larger performance degradation when moving from AES to 3DES than when the IP PSTN architecture was used. The mean values for the CPU times are in Table 8.6.

Heap usage (Figure 8.6), surprisingly, did not follow the same pattern as total and CPU time. While enabling encryption increased the memory used,

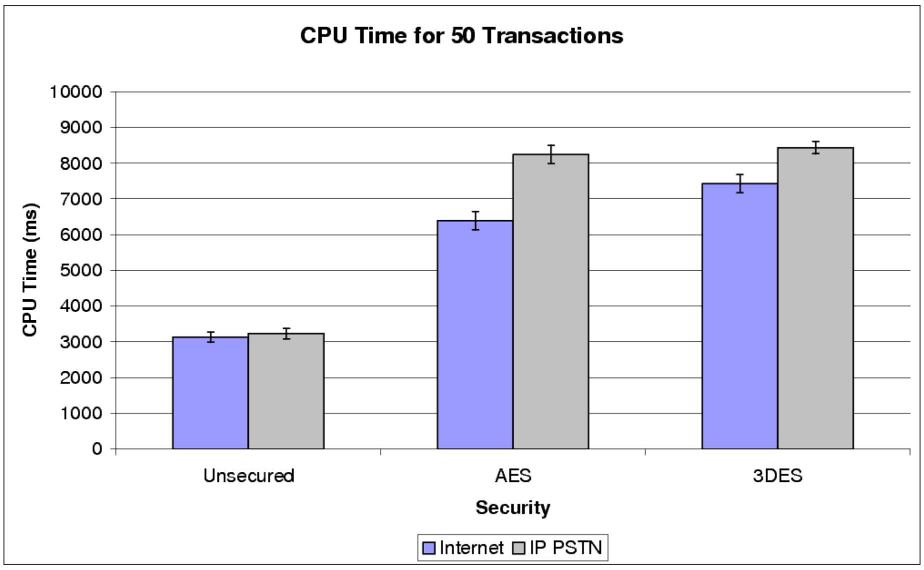


Figure 8.5: Mean client CPU times for 50 transactions.

Security	Internet	S.D.	IP PSTN	S.D.
Unsecured	3128.6	449.5	3227.8	461.9
AES	6391.2	808.0	8242.5	809.9
3DES	7428.5	795.8	8434.9	545.8

Table 8.6: Mean client CPU times for 50 simultaneous transactions. All figures are in ms.

there was no statistically significant difference between the architectures. The relatively larger variance in the encrypted exchanges was likely to be a result of Java's periodic garbage collection.

While there was no difference between the architectures in terms of heap usage, the differences between security schemes were statistically significant ($F_{2,54} = 240.97, p < 0.01$). ANOVA also reported a statistically significant interaction between architecture and encryption ($F_{2,54} = 4.09, p < 0.05$). While statistically significant, it is unlikely to be of practical significance, as the architectures cannot even be statistically differentiated. The mean values for heap usage in this test are in Table 8.7.

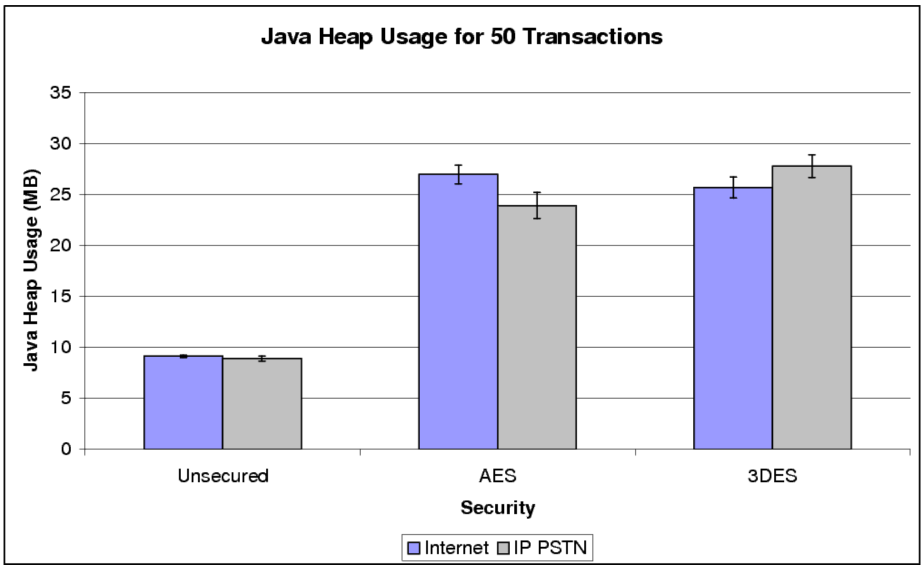


Figure 8.6: Mean client heap usage for 50 transactions.

Security	Internet	S.D.	IP PSTN	S.D.
Unsecured	9.12	0.35	8.90	0.77
AES	26.95	2.87	23.91	4.08
3DES	25.67	3.29	27.77	3.49

Table 8.7: Mean client heap usage for 50 transactions. All figures are in MB.

8.3 Impact of Concentrators

While this experiment measured the total and CPU time as well as memory usage on the client like the previous two, the CPU time and memory usage on the transaction processor was measured as well. The client was configured to send 50 simultaneous transactions using the Internet architecture, with and without concentrators. Figure 8.7 shows the mean CPU time on the transaction processor.

The impact of a concentrator is clear in Figure 8.7; concentrating sessions onto a single session reduced the CPU time on the transaction processor by almost half in the instance of no security, and by over two thirds when security was enabled. Security had significantly less impact on the CPU time required when the sessions were concentrated.

The differences between concentrated and unconcentrated connections was statistically significant ($F_{1,54} = 1041.36, p < 0.01$), as were the differences be-

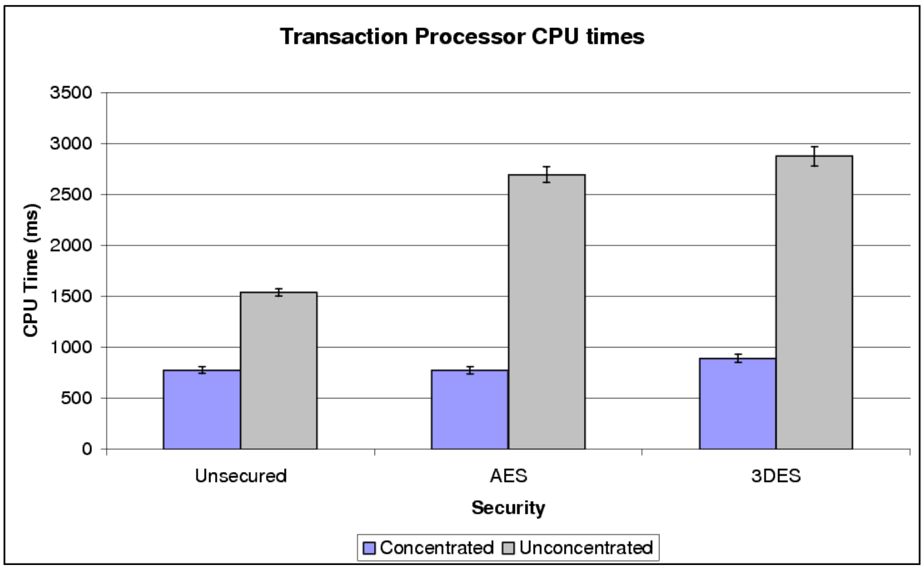


Figure 8.7: Mean Transaction processor CPU times for 50 transactions.

tween the encryption schemes ($F_{2,54} = 84.76, p < 0.01$). The interaction between concentration and encryption was also statistically significant ($F_{2,54} = 67.88, p < 0.01$). This interaction is clear in Figure 8.7; the CPU time required changed very little when security was enabled with concentrated sessions, however, when sessions were not concentrated, the encryption had a significant impact on CPU time used. The mean values for the CPU times are in Table 8.8.

Security	Concentrated	S.D.	Unconcentrated	S.D.
Unsecured	775.6	107.3	1536.9	116.9
AES	775.2	111.4	2693.8	245.9
3DES	892.0	126.1	2873.8	307.8

Table 8.8: Mean transaction processor CPU times for 50 simultaneous transactions using the Internet architecture. All figures are in ms.

Heap usage on the transaction processor is shown in Figure 8.8. Like the CPU times, placing a concentrator in the transaction path had a noticeable effect; memory usage was lower when the sessions were concentrated. This effect was noticeable when the sessions were encrypted.

The differences between the concentrated and unconcentrated sessions was statistically significant ($F_{1,54} = 158.95, p < 0.01$), as were the differences be-

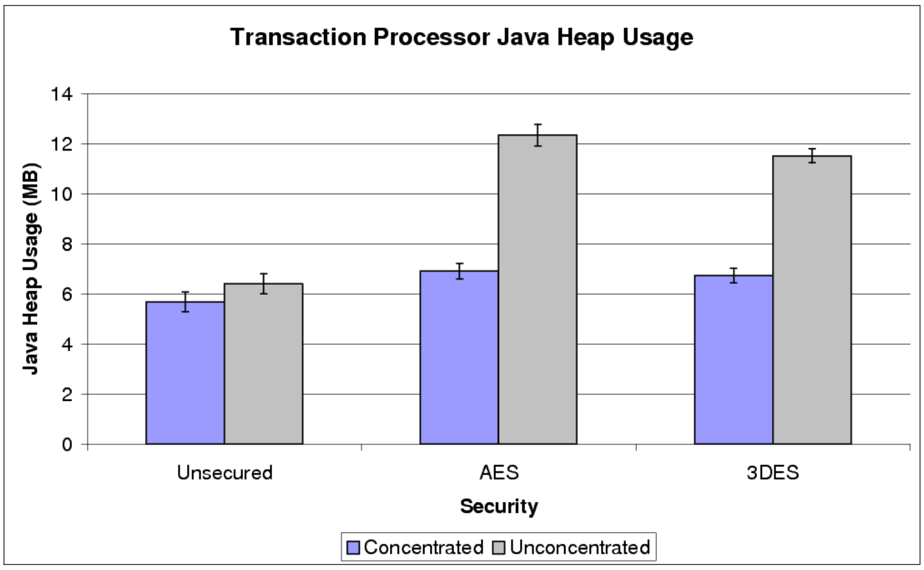


Figure 8.8: Mean transaction processor memory usage for 50 transactions.

tween the encryption schemes ($F_{2,54} = 60.10, p < 0.01$). There was also a statistically significant interaction between concentration and encryption ($F_{2,54} = 26.05, p < 0.01$). This effect was similar to the interaction in the transaction processor CPU times. The heap usage almost doubled when encryption was enabled in the unconcentrated sessions. The same increase in concentrated sessions was significantly less. Table 8.9 contains the mean values.

Security	Concentrated	S.D.	Unconcentrated	S.D.
Unsecured	5.69	1.23	6.41	1.26
AES	6.93	0.99	12.36	1.36
3DES	6.74	0.91	11.53	0.89

Table 8.9: Mean transaction processor memory usage for 50 transactions. All figures are in MB.

Figure 8.9 shows the impact of concentration on the total time required for 50 transactions on the client. The differences between concentrating and not concentrating sessions were not statistically significant. Like in the previous experiments, the differences between the encryption schemes were statistically significant ($F_{2,54} = 132.55, p < 0.01$). There was no statistically significant interaction between concentration and encryption. The mean values for the total times are in Table 8.10.

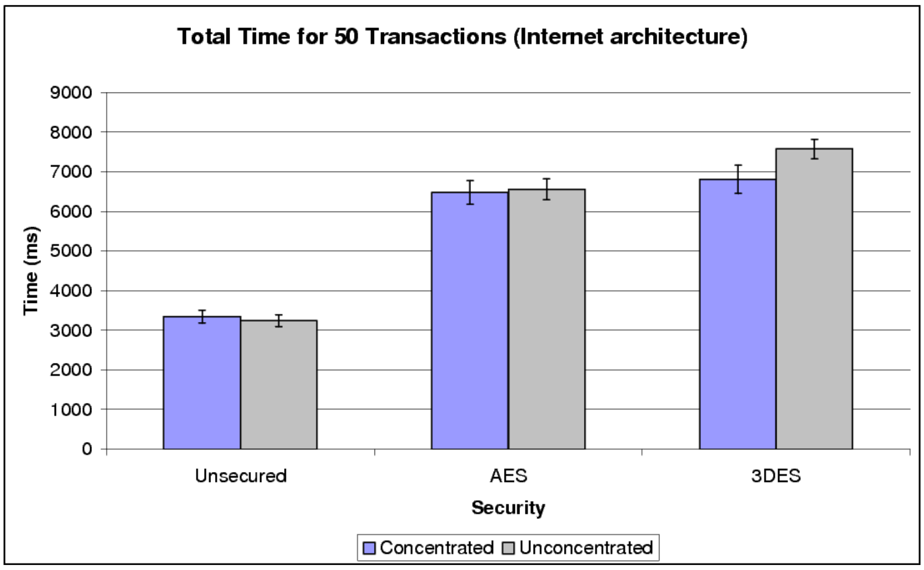


Figure 8.9: Mean times for 50 transactions.

Security	Concentrated	S.D.	Unconcentrated	S.D.
Unsecured	3344.6	515.4	3244.1	460.8
AES	6479.0	940.2	6557.0	818.8
3DES	6806.7	1131.9	7572.5	782.2

Table 8.10: Mean total times for 50 simultaneous transactions using the Internet architecture. All figures are in ms.

Figure 8.10 shows the impact of concentration on the CPU time used by the client. The CPU times echo the total times; the differences between concentrating and not concentrating were not statistically significant. The differences between encryption schemes, however, were statistically significant ($F_{2,54} = 132.41, p < 0.01$). There was no statistically significant interaction between concentration and encryption. The mean values for the CPU times are in Table 8.11.

The effect of concentration on the client's heap usage was similar (Figure 8.11); there were no statistically significant differences between concentrating and not concentrating sessions. Differences between the security schemes were statistically significant ($F_{2,54} = 225.18, p < 0.01$). There was a statistically significant interaction between concentrating and security reported by ANOVA ($F_{2,54} = 3.37, p < 0.05$). However, as the concentrations cannot even be sta-

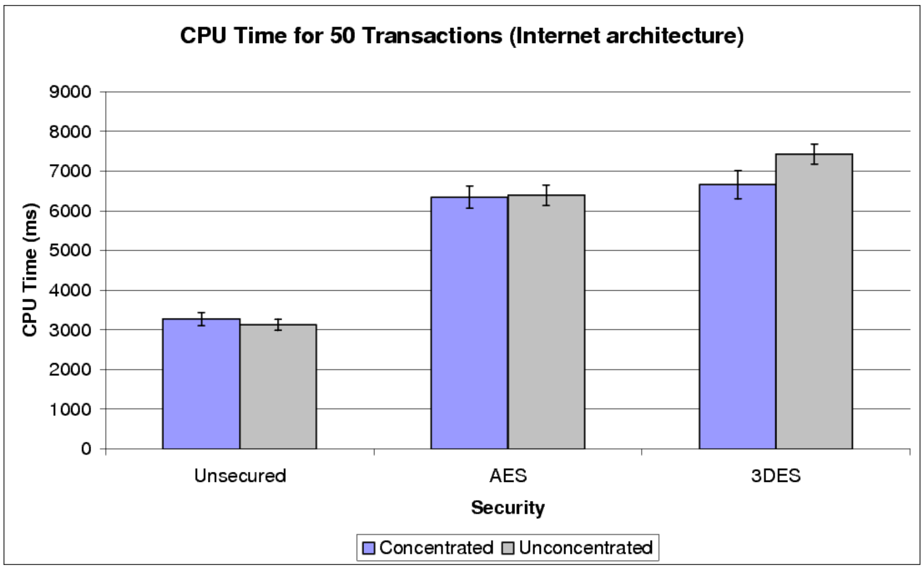


Figure 8.10: Mean client CPU times for 50 transactions.

Security	Concentrated	S.D.	Unconcentrated	S.D.
Unsecured	3273.8	526.7	3128.6	449.5
AES	6343.6	885.3	6391.2	808.0
3DES	6658.0	1124.0	7428.5	795.8

Table 8.11: Mean client CPU times for 50 simultaneous transactions using the Internet architecture. All figures are in ms.

tistically differentiated, the effect of this interaction is small. The mean heap usage for the client are in Table 8.12.

Security	Concentrated	S.D.	Unconcentrated	S.D.
Unsecured	8.96	0.52	9.12	0.35
AES	24.41	4.30	26.95	2.87
3DES	28.00	3.87	25.67	3.29

Table 8.12: Mean client heap usage for 50 transactions. All figures are in MB.

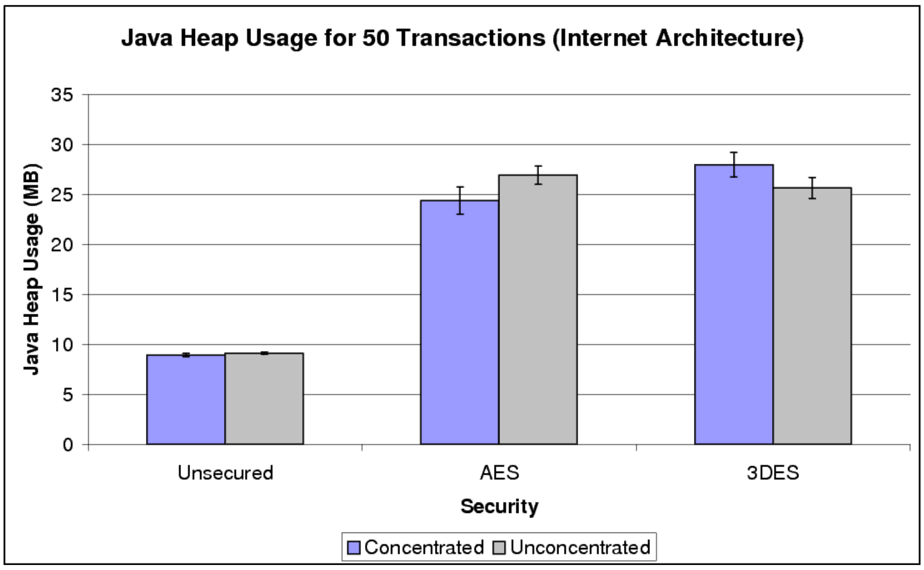


Figure 8.11: Mean client heap usage for 50 transactions.

8.4 Recovery Time

In this experiment, the failure recovery of the system was tested. Two transaction processors registered with the SIP server. The first transaction processor which received the transaction was made to fail after receiving the MESSAGE request from the client. This meant the client had to fallback to the second transaction processor. Figure 8.12 shows the total time required to complete a transaction when a transaction processor failed.

Figure 8.12 shows that the transaction recovery time is highly dependent on the transaction timeout. The total transaction time is simply the time required to start a transaction, the timeout period, then performing the transaction with a working transaction processor. Like the single transaction test, the Internet architecture was slower due to the additional phase.

Time Out	Internet	S.D.	IP PSTN	S.D.
One second	1988.9	61.4	1775.3	19.2
Three seconds	3955.1	31.0	3734.5	47.0
Five seconds	5979.7	61.1	5703.2	110.9

Table 8.13: Mean transaction times for a single transaction with a transaction processor failing mid-transaction. All figures are in ms.

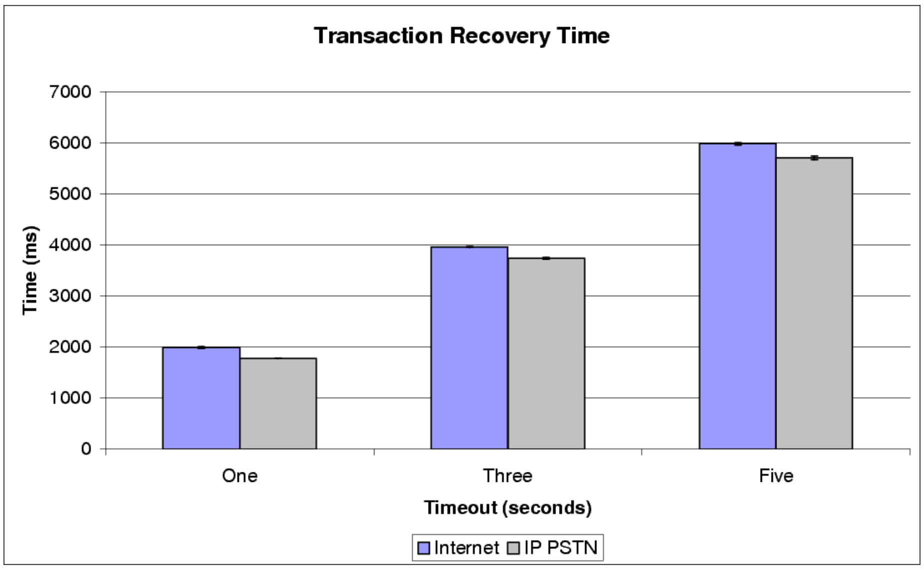


Figure 8.12: Mean transaction time for a single transaction with a transaction processor failing mid-transaction.

The differences between the architectures were statistically significant ($F_{1,54} = 216.35, p < 0.01$) as were the differences between the timeouts ($F_{2,54} = 20144.60, p < 0.01$). There was no interaction between the architecture and delay. Table 8.13 shows the mean values.

Figure 8.13 shows the CPU usage on the client. The CPU times were not affected by the transaction timeout period. Like the single transaction exchange, the Internet architecture required more CPU time. The differences between the architectures were statistically significant ($F_{1,54} = 313.13, p < 0.01$). There was no other significant effect in this test. The values for the mean CPU times used are in Table 8.14.

Time Out	Internet	S.D.	IP PSTN	S.D.
One Second	514.5	33.6	400.5	16.3
Three seconds	515.2	34.1	388.4	18.0
Five Seconds	514.8	33.7	383.2	20.1

Table 8.14: Mean client CPU time for a single transaction with a transaction processor failing mid-transaction.

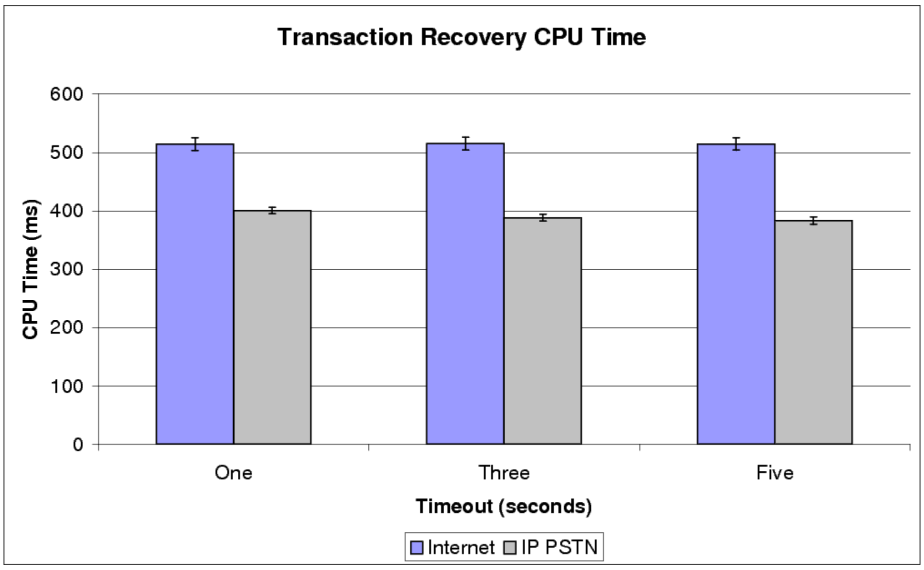


Figure 8.13: Mean client CPU time for a single transaction with a transaction processor failing mid-transaction.

8.5 Discussion

In the single transaction experiments, the Internet architecture was slower than the IP PSTN model. This was because it had the additional phase of populating the server list and negotiating two sessions: one with the SIP server, then another with the transaction processor. In the instance of a single transaction, this had a larger performance impact than having to maintain two sessions at the proxy; consequently, the Internet model placed a smaller load on the SIP proxy.

In the 50 transaction experiments, however, the Internet architecture was faster than the IP PSTN model as the messages did not need to be proxied. The IP PSTN model placed extra load on the SIP server and had additional latency because the messages were proxied. The larger volume of proxied transactions had a greater effect on transaction time than the impact of the Internet model's additional phase of populating the server list. A faster SIP proxy may have alleviated this.

Using concentrators to aggregate multiple SIP sessions onto a single session reduced the load on the transaction processor in terms of CPU time. Doing this improves the scalability and reliability of the transaction processor. Despite the introduction of an additional entity in the transaction path, concentrating sessions had no statistically significant impact on the transaction time, CPU time or heap usage on the client.

In all experiments, enabling encryption increased the transaction time as well as memory usage. AES was generally faster than 3DES. This has been verified in other studies [84, 86, 122], which found that AES provided faster performance than 3DES. Research showed that, as AES was less processor intensive, it had lower energy consumption [98], which is an important factor in battery life in mobile embedded devices. AES had a larger memory footprint than 3DES in the single transaction exchange; it was found that AES code was larger than 3DES code, but a dedicated hardware solution would reduce this [86].

The heap usage on the client in the 50 transaction exchanges (Figures 8.6 and 8.11) was affected differently by the security schemes, architecture and concentration. For instance, Figure 8.6 shows that, when encrypted with AES, the Internet architecture appeared to use more heap space than the IP PSTN architecture. However, this was reversed when the encryption was changed to 3DES. A similar, although opposite, pattern appeared on the client with concentrated/unconcentrated sessions. Despite the appearance of differences, the large variance made any differences statistically insignificant. The large variance was probably due to Java's periodic garbage collection; the heap usage increases until a garbage collection is performed. This made accurate measurements of memory usage difficult.

Recovery time was found to be highly dependent on the transaction timeout. When the timeout was low, the transaction recovered from transaction processor failure faster. However, this test did not examine stress loading on the transaction processors because when the client was run with a relatively large number of threads, for example 50, not all threads timed out according to the specified timeout and instead fell back to the SIP timeout. The standard SIP timeout is approximately 32 seconds, which was considered too long for this project.

These results demonstrate that the proposed solution stack on either network model provides a viable architecture on which to run a transaction network. The slowest transaction time, 50 transactions using 3DES encryption on the IP PSTN model, was 10.6 seconds. This is below the current system's total transaction time of 15 seconds, as noted in Section 2.1.

8.5.1 Single Transactions

When the client was sending a single transaction, the IP PSTN model was faster. This was because the Internet model had the additional step of populating the server list. In the IP PSTN model, the SIP server operated as a proxy and performed this step on behalf of the client. However, Figure 8.1 shows that, without the additional step, the same set of phases in the Internet model was faster than the corresponding phase in the IP PSTN architecture. While the differences between the architectures were small, they may be significant on an embedded device with limited processing ability.

The Internet model also had to negotiate two TLS connections (two TCP connections in the unsecured test): one with the SIP server/registrar and another with the transaction processor. In contrast, a client in the IP PSTN model only needed to negotiate a single TLS connection with the SBC/SIP Proxy. TLS connection negotiation used approximately a third of the total connection time in these instances. This additional phase accounted for the Internet model's higher CPU time and larger heap usage.

In the IP PSTN model, over one third of the time was spent on establishing the single TLS connection. With the Internet model, almost half, and in the instance of encrypted sessions, over two thirds, of the total time was spent establishing the sessions - this consisted of populating the server list and the actual session establishment. In a high volume transaction environment, the benefit of a concentrated, persistent session is clear; a session does not need to be established for each transaction. Rather than having to establish a new session upon each request, a concentrator can forward transactions on the session it maintains with the transaction processor.

This benefit mainly applies to proxies such as SIP servers and concentrators, as the transaction rate from a single transaction terminal is still low, that is, only a single person can use a transaction terminal at a time. If individual transaction processors were to maintain persistent sessions with transaction processors, a significantly larger amount of state stored on a transaction processor would be required.

The transaction phase (the two MESSAGE request and OK response pairs) when encrypted, required almost twice as much time in the IP PSTN architecture than in the Internet architecture. This was due to the proxy behaviour where the messages were handled twice. Thus even if the transaction processor responded immediately, the response may have been delayed and held up at the SIP proxy.

It is interesting to note that the transaction phase was slower when there was no security in the Internet model. The delay between receiving the MESSAGE from the transaction processor and sending the 200 OK response was substantially shorter in the secured exchanges than in the unsecured exchange.

This delay (Delay 1 in Figure 8.14), when secured, had a wide range of values. AES had a mean delay of 36.4ms (s.d. 40.9) and 3DES a mean of 20.6ms (s.d. 24.6). Delay 1 in the unsecured transfers had a mean of 76.7ms (s.d. 5.5).

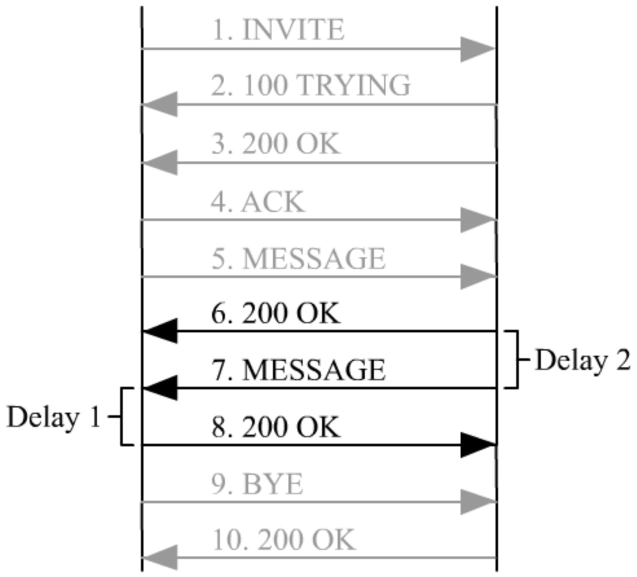


Figure 8.14: Messages 1-4 is the Session Establishment phase, 5-8 is the Transaction Phase and 9 and 10 make up the Session Termination phase.

The Delay 1 difference between the secured and unsecured exchanges was also present in the IP PSTN architecture; unsecured had a mean delay of 76.4ms (s.d. 6.4), AES was 28.3 (s.d. 24.0) and 3DES was 22.9 (s.d. 17.9). As indicated by the large standard deviation, the values for Delay 1 in the secured exchanges also had a wide range of values. However, as the IP PSTN model proxied messages, the delay between receiving the 200 OK and the subsequent MESSAGE request (Delay 2 in Figure 8.14) from the transaction processor was large enough to hide Delay 1 in Figure 8.1. Delay 2, in the IP PSTN model, required almost twice the length of time that the Internet model required. This was because of the IP PSTN's proxying behaviour. The code to handle incoming MESSAGE requests was identical in all tests yet Delay 1 was longer when unencrypted; this is assumed to be an issue with the SIP library.

Terminating the session was similar across all transfers regardless of architecture or whether encryption was enabled. Depending on the volume of transactions, a concentrated, persistent session could also bring benefits here in terms of transaction time; after each transaction, the session will not need to be terminated.

8.5.2 Multiple Transactions

The Internet model was faster than the IP PSTN architecture when the client was made to send 50 transactions, in contrast to the single transaction tests, where the Internet architecture's additional phase caused it to be slower. Here, the load imposed by the additional phase in the Internet model was eclipsed by having to proxy the transaction in the IP PSTN model.

The effect of the proxy was evident in the secured exchanges. The proxy had to maintain two TLS sessions for each client; one to the client and another to the transaction processor. The additional latency of having to handle messages twice also had an effect: CPU time is a smaller proportion of total time in the IP PSTN model. This signified that the client had more idle time in the IP PSTN architecture. This is because of the additional latency of waiting for the request and response to be proxied back.

This did not explain, however, the increased CPU time in the IP PSTN exchanges; the IP PSTN's CPU usage was higher than the corresponding exchange in the Internet model. This increased CPU time was unexpected as the client in the IP PSTN architecture sent fewer messages than in the Internet architecture; thus, fewer messages had to be handled and secured. It was possible that the SIP library did not relinquish the CPU when waiting for a response.

To investigate this, an additional test was run. The aim was to investigate whether the SIP library/JVM really did relinquish the CPU when waiting for a response. The Internet architecture with AES encryption was used. The client was run with 10 threads and the CPU time was recorded. This was repeated 15 times. The transaction processor software was then modified to wait 1000ms before responding to a request. The client was run again with 10 threads. Without the 1000ms delay, the client CPU time was 1220.6ms (s.d. 213.5). When the transaction processor had the delay, the CPU time on the client increased to 1542.2ms (s.d. 265.5). This difference was statistically significant ($t_{28} = 13.37, p < 0.01$). This result showed that, despite exchanging the same messages, when the transaction processor was delayed, the client continued to use a small amount of CPU time when apparently idle.

This explained the increased CPU usage in the IP PSTN model. There was more idle time in the IP PSTN model, due to the proxying behaviour. Idle time contributed a small amount of CPU time, resulting in a higher CPU time. This effect is probably only present when the client is executed with multiple threads; the CPU time used did not change in the recovery tests with different timeouts. This was because the recovery tests were run with only a single thread.

There was a statistically significant interaction between architecture and encryption. The Internet model experienced a noticeable increase in total and client CPU time when switching from AES to 3DES; the IP PSTN model had little difference. As noted above, there was more idle time in the IP PSTN model than in the Internet model. The SIP server was a Pentium 4 2.4GHz, while the transaction processor was an Athlon 800MHz. This suggested the transaction processor was the bottleneck; the SIP server was forwarding and processing requests faster than the transaction processor could handle them. This meant the SIP server and therefore the client were waiting on the transaction processor to return responses. Switching from AES to the more processor intensive 3DES meant less idle time on the SIP server. That is, some of the server's idle time in AES was consumed as CPU time when switching to 3DES, resulting in a similar total transaction time.

Contrast this with the Internet model. The client and transaction processor communicated with each other directly after populating the server list. Any increase in processing time on either end resulted in an overall increase in time. As 3DES is a more processor intensive cipher, client CPU time increased, as indicated in Figure 8.5. Total time increased as well because the transaction processor took longer to respond as a result of the slower cipher.

There was no statistically significant difference between the two architectures in terms of heap usage. This was because with a larger number of threads, a larger proportion of the heap would have been used to store objects; with the single transaction test, a larger proportion of the heap would have been used to store classes. That is, once the classes have been loaded, the cost of storing additional objects is low. This explains that despite sending 50 times more transactions with 50 times more threads, heap usage only increased by half in the unsecured transactions and only approximately tripled in the secured exchanges.

8.5.3 Impact of Concentrators

Concentrating sessions onto a single session to the transaction processor significantly reduced the CPU time and heap space used on the transaction processor. This was expected, as the transaction processor no longer had to deal with incoming sessions, their termination or allocate memory for their maintenance. The transaction processor maintained a single persistent session with the concentrator and only had to deal with the actual MESSAGE requests.

There was a statistically significant interaction between concentration and encryption. When the sessions were concentrated, there was little change in the CPU time required on the transaction processor. Heap usage increased by a relatively small amount as well. However, the CPU time required with unconcentrated sessions increased significantly when encryption was enabled. Memory usage, in the encrypted exchanges, almost doubled. There were several factors which caused this. One was that the transaction processor simply did not need to maintain a session for each client; it only maintained a single session with the concentrator. This reduced the amount of state required by the operating system to maintain network connections. The second was that since sessions did not need to be created and terminated, fewer packets were exchanged. This included the INVITE and BYE sequences. Fewer packets meant the transaction processor did not need to encrypt as much traffic, resulting in less CPU time used.

Concentrating sessions had no performance impact on the client. That is, when the sessions were concentrated, total time, CPU time and heap usage could not be statistically differentiated from the unconcentrated sessions. This was despite having the concentrator positioned in the transaction path.

A concentrator reduces the processing load and memory usage on a transaction processor while serving the same number of clients. This enables the same transaction processor to process transactions from more clients than if sessions were not concentrated. This can reduce costs as well as improve reliability; if the transaction processor is not running at full load, the additional spare processing capacity means it can still process transactions in the event of a sudden jump in transaction rate. These benefits come at no cost to the client.

8.5.4 Recovery Time

Recovery time for a single transaction was found to be highly dependent on the transaction timeout. Total transaction time was approximately 700-1000ms (the normal time required for a single transaction) added onto the transaction timeout value. The Internet architecture's additional phase caused it to have a longer transaction time, resulting in a longer recovery time than when the IP PSTN model was used. When compared to the total transaction time (including the transaction timeout), however, the extra time required for the Internet model's additional phase was small. This test demonstrated the fault tolerance mechanism of the solution stack was indeed viable.

CPU time on the client was unaffected by the transaction timeout value. Like in the single transaction test, the CPU time was higher when the Internet model was used because of the additional phase. CPU usage was higher (when a transaction processor failed) than the CPU times in the single transaction test as the client had to establish a session with a different transaction processor and retransmit the transaction after failure of the first transaction processor.

A test with 50 parallel transactions was not performed. This was because the timeout task associated for each thread did not work correctly on all threads. When this timeout task failed, the corresponding thread would wait for the SIP timeout before timing out (approximately 32 seconds). The standard SIP timeout is defined in RFC 3261 as 64 * the retransmit time (usually 500ms). SIP's retransmit timer, however, is defined for UDP and other unreliable transports, and retransmissions are not performed at the application layer when a reliable transport protocol is used (such as TLS/TCP in this instance). Testing a large number of parallel transactions would have provided details on the performance of the transaction processor when subjected to a heavy load. These results would have been dependent on the hardware used, nevertheless, they still would have had value.

Transaction timeouts when set too low could potentially congest the network. Clients could mistakenly assume a server was down when the cause of the timeout was a latency spike. That is, clients may not be able to distinguish server failure and transaction queuing/processing delays. Further work is required as the transaction timeout must be tuned to the latencies of the network on which the transaction network is deployed. Transaction timeouts would also depend on the processing capacity of the network, that is, the combined transaction processing capacity. Transaction timeouts also need to keep within reasonable levels of responsiveness, for example an EFTPOS terminal must complete the transaction within reasonable time before the user finds the wait too long.

8.5.5 Limitations

There were several major limitations with these experiments, however none of them affected this project's main goal of defining a protocol stack suitable for short duration transactions. Some of these provide avenues for future work.

One assumption was that running 50 threaded transactions on the client machine approximated a processor with 1/50th of the speed. This resulted in the client maintaining 50 connections, instead of a processor 1/50th of the speed maintaining a *single* connection. Therefore, the increased CPU time was not only attributed to encrypting 50 times more data, but also administering 50 times the number of connections and sessions as well. Using an actual embedded device with the processing capability typical of a hand held transaction terminal would have provided the most accurate results. Fortunately, regardless of whether this assumption was valid or not, using 50 threaded sessions provided details of the performance of the SIP server and transaction processor when presented with a large number of simultaneous transactions.

Therefore, a major limitation of this project was the failure to assess the solution stack on an actual embedded device. This would have provided accurate details on the performance of the solution stack and the impact of encryption on an embedded processor. Using an actual embedded device could also have provided specific details such as power usage, memory consumption and code size. However, this would have required a native implementation, which could have increased development time.

Another limitation was that no SBC was actually used, despite it being a crucial part of the IP PSTN architecture. This was because no appropriate SBC was available as explained in Section 6.4. In its place was a SIP proxy server. The SBC would have provided NAT traversal and would have broken the session between the client and the transaction processor; as described in Section 5.2, the clients address messages to the SBC and the SBC forwards these to the destination. SBCs also provide firewall and authentication functionality. None of these were tested in this project. Fortunately these factors were largely external to the function of the protocol stack.

Related to this was the testbed network; it was too simple to represent an actual deployment. In a real scenario, there would be various routers and switches in the path, each adding their own processing and network latency. In addition, there was no real network infrastructure, for example the transaction processor simply returned 300 bytes of data, instead of connecting to a back end database.

While transaction processor failure recovery tests were performed, there was no stress testing of the network and of the transaction processors. This would have provided details on the scalability of the network; while the system works if a transaction processor fails mid-transaction for a single client, there are no guarantees the system will continue to function when there are a large number of simultaneous transactions in the network and a transaction processor fails.

SCTP was also a viable transport protocol for the solution stack. The SIP library in conjunction with the operating system used, however, did not support SCTP.

Finally these results are limited to the hardware used. Faster hardware will give different results. While the absolute values may change, the relative values are expected to be similar; if encryption, for example, requires more time than unsecured exchanges, it is expected that encryption will still require more time than the unsecured exchanges using different hardware.

Chapter 9

Conclusions

This research projected aimed to define a generic IP based protocol stack suitable for short duration transactions and to evaluate the performance of the architectures. Enabling encryption expectedly increased CPU and heap usage as well as time to complete the transaction.

The single transaction experiments approximated a high performance embedded processor. While the differences in time between the two architectures were small, the differences will be significant if: there are a large number of transactions, or embedded devices with limited processing capability are used. Here, the Internet model had slower performance because of the additional phase.

In the multiple transaction tests, the IP PSTN model had slower performance. This was despite the IP PSTN architecture sending fewer messages. This slower performance was due to the proxying behaviour; messages were handled twice, introducing more latency and processing time.

Using concentrators to aggregate multiple sessions onto a single session had a significant impact on the transaction processor. When concentrated, the transaction processor had lower CPU and heap usage compared with unconcentrated sessions. The cost of encryption was also considerably lower on the transaction processor when the sessions were concentrated. Although the concentrator represented another entity in the network path and subsequently introduced more latency and processing time, this had no effect on the total transaction or CPU time used by the client.

Recovery time was found to be highly dependent on the transaction timeout value. Further work needs to be completed to analyse the network availability and reliability under heavy load conditions. This test, however, still demonstrated that the solution stack was robust in the event of a transaction processor failure.

Further work includes assessing the solution stack on an actual embedded device, to attain an indication of real world performance. A more complex and comprehensive testbed network could also be used, to provide a more realistic model of performance. This includes using an SBC appliance, back end databases and additional routers and switches to introduce realistic latency typical of WANs. While TCP was the only transport protocol used, SCTP was also a candidate transport protocol. Measuring the performance using SCTP in place of TCP as well as the additional reliability provided by multihoming is another possibility. With wireless networks becoming more pervasive, work could also be performed on the feasibility of wireless payment networks and the security challenges introduced.

The proposed SIP based solution has been demonstrated to be a viable protocol stack on both the open Internet and the controlled private network model. Both architectures support transactions within reasonable time and demonstrate the ability to recover in the event of transaction processor failure.

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Appendix A

Generating Certificates

OpenSSL and Java's Keytool were used to generate the public/private key pairs.

To generate the CA certificate and private key:

```
openssl req -x509 -new -out ca/cacert.crt -keyout ca/cakey.key -days
9999
```

To generate the SIP server's certificate request and private key:

```
openssl req -config openssl.cnf -out proxy.csr -pubkey -new -keyout
privkey.pem -outform PEM -nodes
```

To sign the SIP server's certificate request:

```
openssl x509 -req -in proxy.csr -CA ca/cacert.crt -CAkey ca/cakey.key
-CAcreateserial -outform PEM -out proxy.pem -days 9999
```

To generate the client's public/private key pair:

```
keytool -genkey -alias clientKey -validity 9999 -keystore clientKey
-keyalg RSA
```

To export the client's certificate signing request:

```
keytool -certreq -alias clientkey -keystore clientKey -file clientKey.csr
```

To import the CA's certificate into the client's keystore:

```
keytool -import -keystore clientKey -file cacert.crt -alias ca
```

Signing this request is the same as for the proxy.

Importing this signed certificate is the same as importing the CA's certificate.

Generating the transaction processor's keys require the same steps as generating certificates for the client.