**Team 17 Methodology, Anonychat**

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In order to accomplish our goal we have designed a messaging protocol that will provide anonymity through the use of a distributed network of peers and multiple layers of encryption. In order to test our design we have implemented a test platform in C for Linux. This test application will allow us to perform testing on our design to ensure that it meets our goals and does not have a huge performance overhead. The details of our design and testing methods will be discussed in detail below.

The Anonychat messaging protocol will consist of a set of distributed clients to act as peers in transmitting messages, as well as a centralized name server to inform clients of available peers. These clients will send messages amongst each other using controlled flooding, encryption between the source and destination peers, and encryption between each intermediate peer. In order for this to work, we need a way to ensure that each peer has a connection to all of the other peers. This is the role of our name server; it will distribute peer lists to each peer, and make sure that they can all communicate.

The Anonychat Name Server (ANS) is responsible for creating a network of peers that provides each peer with a route to each other. Our protocol does not require a specific implementation for creating this network, but there are certain conditions that must remain true. The peer network must be changed periodically allowing peers to send messages on different paths (The peer network must assign peers in a random fashion, for example peers with similar IP address prefixes, or latency should not be assigned to each other.) As stated above, the peer network must not isolate any peers or group of peers; they must all have a route to each other. The ANS must also maintain connection state on all peers currently in the network, and be able to react accordingly when a peer drops from the network.

In our test implementation we maintained state by requiring each peer to uphold a TCP connection with the server. This allowed us both to determine when a peer has connected/disconnected and to send out peer updates to the peer when required. Our node network was implemented using a connection graph, which is detailed below:

1. An internal list of clients is re-arranged to occur in a random order. This should ensure that connections appear random as well for the following steps.
2. Based on a series of internal parameters, such as minimum number of connections and availability of clients, nodes are arranged so that they have a bidirectional connection in the shape of a ring, with an arrangement of connections occurring to cross the ring. If the requested minimum number of connections would mean that some connections would overlap, every node will connect to each node once.
3. When the central server’s graph is completed, a message is sent to every node informing it of its assigned peers. No node is informed of who is assigned to send to it, and the bidirectionality of the connections is only implied by the specific graphing implementation used. Future name servers can be made with differing graphing mechanisms. Additionally, nodes are not to be informed in any way of connections beyond one hop. As a result, attempts to send to a node that is no longer available will simply die in the network, possibly never reaching the target. This is by design, as it means that if no one has the requisite keys to read a particular message, then the message eventually fades.
4. This cycle will repeat either (1) every time a client joins or leaves if the number of clients in the network is less than a set number, or (2) every 30 seconds if there are a larger number of clients. The second case implies that, in larger networks, new clients may have to wait up to 30 seconds before it is able to broadcast messages, but this should also keep the name server from flooding the network. Messages in transit when connections are re-assigned behave as if nothing happened.

The client component or peer of our messaging platform is slightly more complicated. This is because a theoretical adversary with a pervasive view of the network would be able to identify the first broadcast of a message, and from there determine who sent it. In order to alleviate this problem, we turn to an approach similar to onion routing. In onion routing, messages are encrypted with multiple layers of encryption and decrypted layer by layer as the message is bounced around the network though different onion routers. Inspired by this, we use a simpler approach. Our protocol implements two levels of encryption in order to communicate the messages while maintaining both anonymity and confidentiality. Both levels of encryption will use asymmetric encryption, implemented by the RSA algorithm. The first level of encryption, between the destination and source, guarantees that the destination knows the message is for itself, without the source knowing any information about it other than its public key. The second level of encryption ensures that the source and destination of the messages remain anonymous. This level is between each intermediate peer, and ensures that the signature of the message changes as it is sent across the network. However, the content of the message remains the same. The end effect is similar to onion routing; it makes it harder for an external omnipresent observer to trace message senders in the network.

Peers will communicate with each other via controlled flooding. Specifically, they will communicate by broadcasting the message to all of their peers, and their peers will then broadcast the message to their peers and so on. When a peer receives a message that it has never seen before, it will always broadcast the message to its peers, regardless of whether it was the intended destination or not. To determine if a received message is new, a hash of the source/destination encrypted message can be compared against a hash of previously received messages.

Thus far, our implementation is still vulnerable to an adversary that has a view of the entire network as a whole. As previously stated, all the users share their public keys when they connect to the network, and anytime a user sends a message, it is encrypted with the user’s public key. However, as the message is sent between nodes, each node also encrypts the message with the next hop’s public key, which is generated on connection and destroyed on disconnect. This constitutes a second layer of encryption, with

To verify that our protocol is viable, has sufficient performance, and works as intended, we plan to conduct several tests. These tests include measuring message latency, latency introduced by encryption, time differences in successful and unsuccessful decryption, and the network utilization of our controlled flooding implementation. These experiments will be run via a series of simulations on both a local machine and four computers spread out on different networks. The latter is a necessity to get accurate latency results.

We expect that the indirect method our protocol uses to transmit messages will incur additional round trip time (RTT). However, we don’t want this additional latency to be much bigger than the latency found in direct communications. To ensure our latency is acceptable, we will set up our test implementation on our four test machines. We will then send a series of 25 messages between each machine using our messaging protocol, and record the RTT of each message. Next, we will send another 25 messages between the each of the four hosts themselves using ping, and again record the RTT of each message. After collecting the RTT of both messages, we will compare them and verify that the RTT using our protocol is not greater than five times the direct RTT.

Encryption plays a big part in our messaging protocol, but it can also be a very expensive computation. For the sake of rapid communication, we don’t want encryption to add a huge overhead to our protocol. To ensure this is not the case, we plan to test the time it takes for messages of various different sizes to be encrypted and then decrypted. We will send messages with lengths in a range starting from 20 to 200, and increasing by increments of 20 each iteration. Using these tests, we will calculate the time each of these messages took to encrypt and decrypt. This will add additional latency to message sending, but hopefully it will be less than half of the network latency.

In addition to testing the overhead added by encryption, we also need to test that the time difference between a successful and unsuccessful decryption is not noticeable. If this difference is noticeable it could allow an adversary to determine if a particular node was the intended recipient of a message. To do this we will perform a similar test to our total encryption/decryption overhead test mentioned above. However, this time we will decrypt the message twice: once with the correct private key, and the second time with an invalid private key. Comparing these values will allow us to determine if a successful decryption is noticeable.

The network utilization of our protocol also needs to be tested. We don’t want our protocol to use a vast majority of the available bandwidth by flooding messages across the network. In order to test our utilization we plan to run a simulation with around 25 clients running locally. The simulation will last for five minutes with each client periodically sending messages to other clients. To do this, each client will pick a random time from five to 35 seconds. The client will then send the message after this time expires, and then repeat the process. As this occurs we will analyze the network using Wireshark, and take note of how many packets are in the network for 5 second intervals. We will also run a five minute simulation of normal computer use, such as web browsing, and heavy network use such as a file transfer. This will allow us to compare our protocol’s network usage against these two, giving us a sense of how our network utilization compares to normal and heavy network use.

As a chat program could possibly be used by at least 50 people at a time, it seems paramount we test how well our messaging protocol will scale. To do this we will perform a test similar to the network utilization test; however, instead of analyzing the network, we will be analyzing the resource usage of each node and the ANS. During this simulation we will record the number of messages each node processes, and the processing time of each of those messages. We will also record the RTT of each message to determine the effect more clients has on it. We will also be analyzing the ANS, measuring how many times it reshuffles the peer list and how long each of those peer reshuffles took. Unlike the network utilization test, we will run this test on a range of clients from ten to 30, increasing by ten clients each time. This will give us a clear measure of how well our protocol scales with varying numbers of clients.

The final test that we will perform is the anonymity test of the source of our messages. To do this, we will run a simulation with around ten clients sending messages back and forth to each other at varying intervals. We will use wireshark to examine the sent packets, and take note of whether the packets are encrypted as well as confirm there are no identical packets sent between peers. This will help us tell if it is possible to determine the source of a particular message by analyzing the packets sent between nodes.

We hope that these tests will provide us with an overall measure of how well that our protocol is performing, and how well we have performed at reaching our goals.