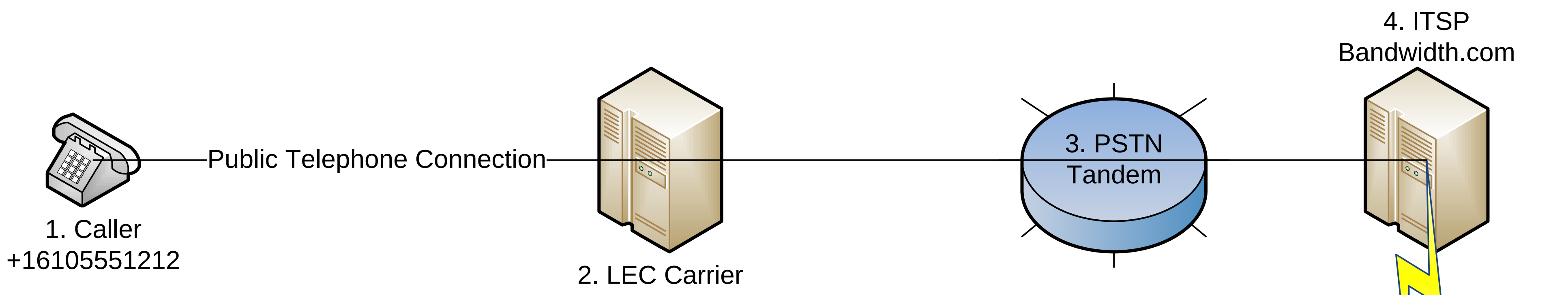

1. Caller dials 14345551212, goes to their TELCO (2).

2. Telco looks up number and sends it via Tandem (interconnect/LD switch) to ITSP (4).

3. ISTP (bandwidth.com sees the number, says it bleongs to customer on IP 65.66.67.68 over the Internet (5) and sends the invite to port 5080 udp.

4. pfSense (6) firewall sees this and forwards it via symmetric NAT to sipXecs at IP 192.168.2.10.

5. sipXbridge (7) sees this media relay sends the invite to sipXecs (7) on port 5060. sipXecs says the DID belongs to user 200, who is registered and the invite goes to the phone (8).

6. User 200 answers the call and audio is established on port 30000 inbound/outbound.

The next concurrent calls uses the next port, etc. That's why symmetric NAT is important on your firewall. The outbound port and inbound port has to remain the same and not be rewritten by your firewall.