Name . . . . ID . . . . Section . . . . . . Seat No . . . . . .

Sirindhorn International Institute of Technology
Thammasat University

Final Exam Answers: Semester 2, 2011

Course Title: ITS413 Internet Technologies and Applications

Instructor: Steven Gordon

Date/Time: Thursday 12 April 2012; 9:00–12:00

Instructions:

- This examination paper has 16 pages (including this page).
- Conditions of Examination: Closed book; No dictionary; Non-programmable calculator is allowed
- Students are not allowed to be out of the exam room during examination. Going to the restroom may result in score deduction.
- Students are not allowed to have communication devices (e.g. mobile phone) in their possession.
- Write your name, student ID, section, and seat number clearly on the front page of the exam, and on any separate sheets (if they exist).
- Reference material at the end of the exam may be used.

Internet Technologies and Applications, Semester 2, 2011

Prepared by Steven Gordon on 14 April 2012
ITS413Y11S2E02, Steve/Courses/2011/S2/ITS413/Assessment/Final-Exam.tex, r2302

Question 1 [8 marks]

Explain the following delivery mechanisms and give an example application (or application protocol) that is commonly used by the mechanism.

(a) Unicast [2 marks]

Answer. *Send from a single source to a single destination. Example: web browser (client to server).*

(b) Broadcast [2 marks]

Answer. *Send from a single source to all hosts in a network. Example network management (DHCP).*

(c) Multicast [2 marks]

Answer. *Send from a single source to selected hosts in a network. Example: video streaming*

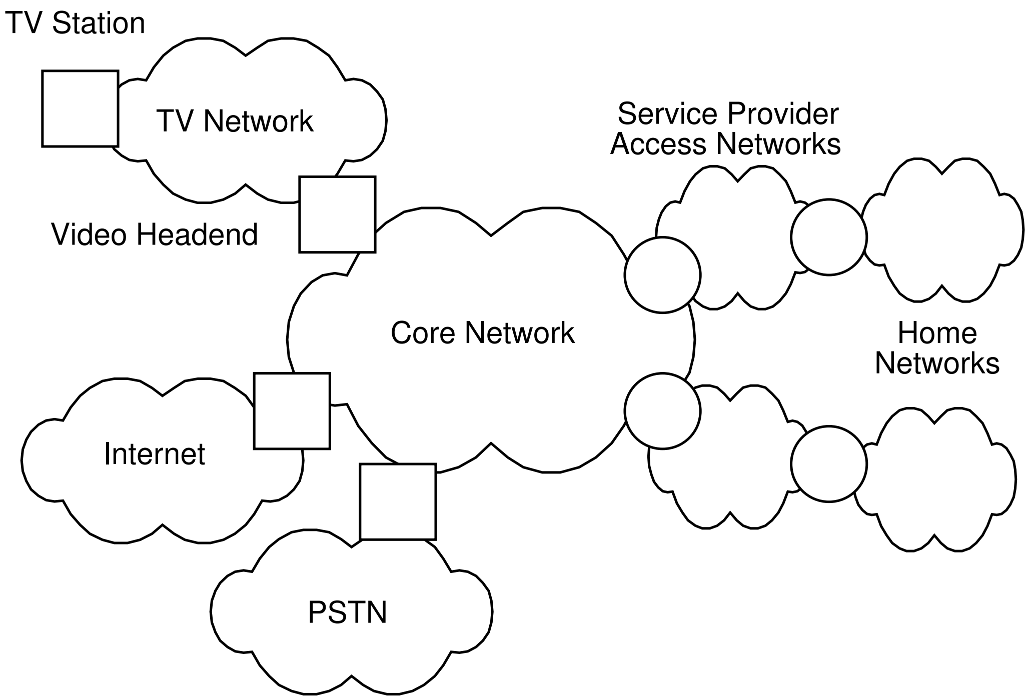
(d) Anycast [2 marks]

Answer. *Send from a single source to any one of a selected group of hosts. Example: DNS*

Question 2 [10 marks]

Consider an IPTV network operated by a single network operator.

(a) Draw a diagram illustrating the network topology of the entire network, showing at least: core network, service provider access network, home network, PSTN, Internet, TV network. Also identify the video headend, and two subscribers on separate access networks. [4 marks]



(b) Explain the difference between ADSL2+, FTTH and FTTN, as options for the service provider access network in an IPTV network. State the transmission media they use and an advantage the technology has (compared to the other two). [3 marks]

i. ADSL2+

Answer. *ADSL2+ uses copper line to the home from a telephone exchange serving a small suburb. The advantage of ADSL2+ is that it uses existing telephone networks.*

ii. FTTH

Answer. *FTTH delivers optical fibre direct to a home, avoiding copper lines. The advantage of ADSL2+ is that it uses existing telephone networks. FTTH delivers the highest speed but at the highest of installing optical fibre all the way to each home.*

iii. FTTN

Answer. *FTTN uses optical fibre to a special node that is closer to homes, then uses copper to the homes. It serves 100's of users. FTTN increases speeds (because the copper link is shorter) compared to ADSL2+ but requires extra cost for the nodes.*

(c) Explain why and how normal TV and video-on-demand are treated differently when delivered across a service providers core network. [2 marks]

Answer. *With normal TV each channel is viewed by many people at the same time. Therefore the content can be delivered using multicast across the core network, rather than using multiple unicast streams to each subscriber. With video-on-demand it is unlikely multiple subscribers will watch the same video at the same time, and therefore multicast cannot be used—a separate stream is needed for each user.*

(d) What protocol should be used for a subscriber to change channels when viewing normal TV across the IPTV network? Explain your answer. [1 mark]

Answer. *IGMP*

Question 3 [8 marks]

Consider a national IP network owned and operated by a single operator (ISP).

(a) Explain the difference between soft QoS and hard QoS. [2 marks]

Answer. *Soft QoS allows priority to be given to specific packets, but cannot guarantee applications/hosts/packets will receive an absolute performance guarantee, e.g. never less than 1Mb/s. Hard QoS provides performance guarantees, e.g. an application is guaranteed to receive at least 1Mb/s.*

Assume packets coming from hosts are classified and marked at the first operator-owned router they arrive at (such a router on the edge of the operators network is called an *edge router*). The packets are then sent across the operators core network to the eventual destination hosts. Routers inside the core network apply QoS control on the packets according to their marking(s).

(b) Explain how an edge router can classify packets, referring to specific protocols and header fields. [2 marks]

Answer. *Packet classification determines which packets are treated in which manner. Packets are normally classifed by users and/or applications. Users can be determined by source and destination IP addresses. Applications can be determined by transport protocol, and source and destination port numbers in the TCP/UDP header.*

(c) Explain how packets may be marked by the edge router, referring to specific protocols and header fields. [2 marks]

Answer. *Packet marking sets the packet to receive a certain QoS based on packet classification. The marking is read by subsequent routers. The DiffServ or ToS field in the IP header can be used for packet marking.*

(d) A router can provide priority to packets in or entering its transmit (output) queue using a forwarding or *queuing scheme* and a *dropping scheme*. Explain the difference between the two. [2 marks]

Answer. *A queuing scheme determines which packets in the transmit queue are transmitted first. A dropping scheme determines which packets are dropped first.*

Question 4 [13 marks]

Consider a university using an IP network to stream black and white video from various security cameras back to a central office (with security guards and recording equipment). There is no audio recorded. Each camera has a dedicated 100Mb/s Ethernet link to one of many 24-port site switches. Each site switch has a 100Mb/s Ethernet link to a 48-port switch in the central office. The switch in the central office has a 100Mb/s Ethernet link to a server that records and displays the video. Cameras record video at a resolution of 1024x768 and frame rate of 1 frame per second; they stream raw black and white video (no compression is used). RTP is used to transport the video: the maximum payload in each RTP packet is 128 Bytes. Assume the Ethernet data link and physical layers contribute 12 Bytes of header per frame. RTSP is used to control the cameras (e.g. a user in the central office sends a message to start the camera streaming).

(a) How many security cameras can be supported on the network? [6 marks]

Answer. *With a resolution of 1024x768 and 1fps the camera sends 786,432 bits per second. Note that the image is black and white, so just 1 bit per pixel. Each RTP packet contains 128 Bytes or 1024 bits of payload. That means 768 RTP packets are sent per second. Each RTP packet has a 12 Byte header, plus the 8 Byte UDP header, plus the 20 Byte UDP header, plus the 12 Byte DLL/PHY header. Therefore each RTP packet contains 180 Bytes. A camera transmits 768 packets per second with 180 Bytes per packet giving a total of 1,105,920 bps. The bottleneck link is from server to central office switch: 100Mb/s. Therefore 90 cameras can be supported.*

(b) Although the security cameras do not support compression, the server can compress the received video before saving to a hard disk. There is one 10GB hard disk allocated for each camera. The compression reduces the file size by a factor of 100. How many hours of video can be recorded for a single security camera? [2 marks]

Answer. *A single camera generates 786,432 bps. With a compression factor of 100, the server needs to save 7,864.32 bps per second. That is 28.311552 Mb per hour or 3.538944 MB per hour. With a 10GB hard disk that is 2825 hours of video.*

(c) Consider the network connecting the cameras to the central office and server. The cameras and site switches cannot be upgraded, but other equipment can be. Explain how you could modify the network to support more cameras. [2 marks]

Answer. *The bottleneck is the link from the central office switch to the server. To increase the capacity of the bottleneck you can upgrade to 1Gb/s, or bind multiple 100Mb/s links together and connect to multiple LAN cards on the server, or run multiple parallel recording servers.*

Some applications/services use different protocols for sending data compared to controlling the data transfer (such as controlling streaming sessions). The different protocol stacks can be divided into the *data plane* and *control plane*.

(d) Draw two protocol stacks for the video streaming application on the camera: one for the *data plane* and the other for the *control plane*. [3 marks]

Answer. *App*

RTP

TCP/UDP

IP

Ethernet DLL

Ethernet PHY

App

RTSP

TCP/UDP

IP

Ethernet DLL

Ethernet PHY

Question 5 [6 marks]

Consider the times at which six packets were transmitted by a source and received by the destination (all times are relative to an initial clock value and measured in milliseconds (ms); the clocks at source and destination are synchronised):

Packet 1 Transmit time: 20; Receive time: 42

Packet 2 Transmit time: 40; Receive time: 62

Packet 3 Transmit time: 60; Receive time: 80

Packet 4 Transmit time: 80; Receive time: 103

Packet 5 Transmit time: 100; Receive time: 119

Packet 6 Transmit time: 120; Receive time: 140

(a) What is the average delay experienced in the network? [1 mark]

Answer. *Average of: 22, 22, 20, 23, 19, 20 = 21ms*

(b) What is the jitter experienced in the network? [1 mark]

Answer. *Average of: 0, 2, 3, 4, 1 = 2ms*

(c) Playback buffers are often used to compensate for jitter. Explain how a playback buffer can be used in this case, and how it reduces the effect of jitter. [1 mark]

Answer. *When a packet is received, it is buffered before played at the receiver. The time of buffering is such that the playback occurs at a regular interval.*

(d) Using a playback buffer, what is the preferred playback time of each of the 6 packets? [2 marks]

Answer. *43, 63, 83, 103, 123, 143.*

(e) One disadvantage of playback buffers is the additional complexity/memory needed. What is another disadvantage of using a playback buffer? Use the example six packets to explain. [1 mark]

Answer. *An additional delay is introduced before playback starts. In this example, playback starts at time 43 (delay of 23ms), as opposed 40 (delay of 20ms), i.e. an extra 3ms delay.*

Question 6 [8 marks]

(a) Explain the role of an indexer in a Bittorrent network. [1 mark]

Answer. *An indexer maintains a list of .torrent files and associated descriptive information. Used for searching for .torrent files.*

(b) Explain the role of a tracker in a Bittorrent network. [1 mark]

Answer. *A tracker manages the set of peers in a swarm accessing a torrent. It maintains a list of peers in the swarm and statistics about the swarm.*

(c) What application protocol does a Bittorrent client use to communicate with a tracker? [1 mark]

Answer. *HTTP*

Assume a peer, N_1, has joined a swarm and established two connections to peers N_2 and N_3. Peer N_1 wants to download a torrent with 100 pieces: $P_1, P_2, P_3, \ldots, P_{100}$. Each piece has 10 blocks (e.g. piece P_1 has blocks $B_{1,1}, B_{1,2}, \ldots, B_{1,10}$; piece P_2 has blocks $B_{2,1}, B_{2,2}, \ldots, B_{2,10}$). Peers N_2 and N_3 already have the following pieces:

N_2**:** $P_5, P_{10}, P_{11}, P_{13}, P_{20}, P_{23}, P_{30}, P_{39}, P_{64}$

N_3**:** All pieces *except* pieces P_5, P_{20}, P_{30}

In the Peer Exchange Protocol, after an initial *Handshake*, each peer exchanges a *Bitfield* message which indicates the pieces they have available. Then a peer may send a *Request* message to request a specific block, and receive a *Piece* message containing a specific block.

(d) Assuming only peers N_1, N_2 and N_3 are in the swarm, what is the availability of the torrent? Explain you answer. [1 mark]

Answer. *The availability is 1.06; of the 100 pieces in the torrent, all 100 pieces are available, and there are 6 pieces that have two copies*

(e) If there were another two seed peers in the swarm, N_4 and N_5, what would the availability be? [1 mark]

Answer. *As they are seed peers, they both have all 100 pieces. The availability will be 3.06.*

(f) In the *Bitfield* message sent from N_2 to N_1, what values will be included? [1 mark]

Answer. *The list of pieces that* N_2 *has:* $P_5, P_{10}, P_{11}, P_{13}, P_{20}, P_{23}, P_{30}, P_{39}, P_{64}$

Each peer maintains four variables for each other peer it is connected to: *am_choking*, *am_interested*, *peer_choking*, *peer_interested*. Consider the values that peer N_1 maintains for the other two peers:

N_2**:** *am_choking=False*, *am_interested=True*, *peer_choking=False*, *peer_interested=True*

N_3**:** *am_choking=False*, *am_interested=True*, *peer_choking=True*, *peer_interested=False*

(g) Will N_3 send a *Request* message to N_1? Explain your answer. [1 mark]

Answer. *No, because* N_3 *is not interested in pieces that* N_1 *has.*

(h) Can N_1 download pieces from N_3? Explain your answer. [1 mark]

Answer. *No, because* N_3 *is choking* N_1*.*

Question 7 [8 marks]

Consider the following scenarios of TCP sessions and give the expected throughput for the specified session. Explain each answer.

(a) Application A on computer 1 sending data to application B on computer 2. Computers connected via network with RTT of 10ms and bottleneck capacity of 20MB/s. The TCP socket used by application A has a send buffer size of 1MB, while the TCP socket used by application B has a receive buffer size of 100KB. Ignore the impact of congestion control (no packet losses). What is the approximate throughput for which A can send to B? [2.5 marks]

Answer. *The bandwidth-delay product is 200KB. The receive buffer (100KB) is less and therefore is the limit on the throughput. Application A can send at a maximum rate of 100KB per RTT or 10MB/s.*

(b) Same as part (a), except the RTT is 5ms, the send buffer size is 2MB and the receive buffer size is 200KB. [2.5 marks]

Answer. *The bandwidth-delay product is now 100KB. The receive buffer (200KB) is greater and therefore the throughput is limited by the bottleneck. Application A can send at 20MB/s.*

(c) Application A on computer 1 sending data to application B on computer 2 using three TCP connections (RTT of each connection is 10ms). Application C on computer 3 sending data to application D on computer 4 using two TCP connections (RTT of each connection is 10ms). Computers 1 and 3 are on the same subnet; computers 2 and 4 are on the same subnet. The path between the two subnets has a bottleneck capacity of 20MB/s. What is the approximate throughput for which A can send data to B? [3 marks]

Answer. *There are five TCP connections sharing the same bottleneck link. TCP is fair and so each TCP connection obtains approximately an equal share of the bottleneck capacity, i.e. 4MB/s. Application A has three connections and therefore achieves a throughput of 12MB/s.*

Question 8 [8 marks]

Consider the three P2P systems: Napster, Gnutella and Fasttrack. For the following features, indicate which P2P system is best characterised by the feature. Either write the name of only one of the three systems ("Napster", "Gnutella" or "Fasttrack") or write the word "None" to indicate none of the three P2P systems have that feature.

(a) Uses a central index server *Napster*

(b) Super-peers *Fasttrack*

(c) Single point of failure *Napster*

(d) Fully distributed *Gnutella*

(e) Search queries are flooded *Gnutella*

(f) Selected peers store index data *Fasttrack*

(g) Content stored on central server *None*

(h) Fastest search *Napster*

Question 9 [11 marks]

Consider a P2P system using a Chord-based Distributed Hash Table. There are a maximum of 32 peers in the system. There are currently 10 peers in the system, with IDs:

1, 4, 10, 12, 15, 19, 21, 23, 24, 29

There are 13 resources that have already been inserted into the system. Their keys are:

0, 1, 4, 7, 9, 13, 15, 18, 22, 23, 24, 27, 28

(a) What is the length of the hash value used in this P2P system? [1 mark]

Answer. *5 bits, since the maximum number of peers is 32.*

(b) Which peer stores the resource with key [2 marks]:

i. 15? *15*

ii. 27? *29*

(c) Which other peers does peer 1 have routes to? [3 marks]

Answer. *Peer 1 would ideally have routes to peers 1, 2, 4, 8 and 16 positions away, i.e. peers 2, 3, 5, 9, and 17. However if such a peer doesn't exist, the nearest successor is chosen. So peer 1 has routes to peers 4, 10 and 19. In more detail, the routing table for peer 1 contains:*

- *1 position away (2): peer 4, key space 2*
- *2 positions away (3): peer 4, key space 3—4*
- *4 positions away (5): peer 10, key space 5—8*
- *8 positions away (9): peer 10, key space 9—16*
- *16 positions away (17): peer 19, key space 17—0*

(d) If peer 1 is searching for resource with key 27, which peers is the search query message sent via to reach the destination? [2 marks]

Answer. *From the question above, to find the peer with key 27 peer 1 will send to peer 19. Then by find the routing table for peer 19 (see below), the query will be sent to peer 29. Peer 29 manages key 27 and hence the query has reached the destination. The routing table for peer 19 contains:*

- *1 position away (20): peer 21, key space 20*
- *2 positions away (21): peer 21, key space 21—22*
- *4 positions away (23): peer 23, key space 23—26*
- *8 positions away (27): peer 29, key space 27—2*
- *16 positions away (3): peer 4, key space 3—18*

Consider the same Chord-based DHT in use in a larger P2P system with more than 100,000 nodes. An alternative is to use Fasttrack (instead of Chord), with 100 super-peers.

(e) Compare Chord against Fasttrack in terms of search overhead (i.e. the number of copies of single query sent in the network). [1.5 marks]

Answer. *For Chord the maximum number of queries sent is 17. For Fasttrack the minimum number of queries is 1 when the resource is on a peers super-peer. But if not, broadcasting between the super-peers is needed meaning the number of queries grows rapidly (often much larger than 17). On average Chord will have less messages than Fasttrack.*

(f) Compare Chord against Fasttrack in terms of failure resistance. [1.5 marks]

Answer. *For Fasttrack, the failure of a normal peer has little impact, but the failure of a super-peer impacts on 1000 normal peers. For Chord the failure of a single peer impacts only on several peers (successor needs to update keys, routing tables may need to be updated). Chord is more failure resistant.*

Reference Material

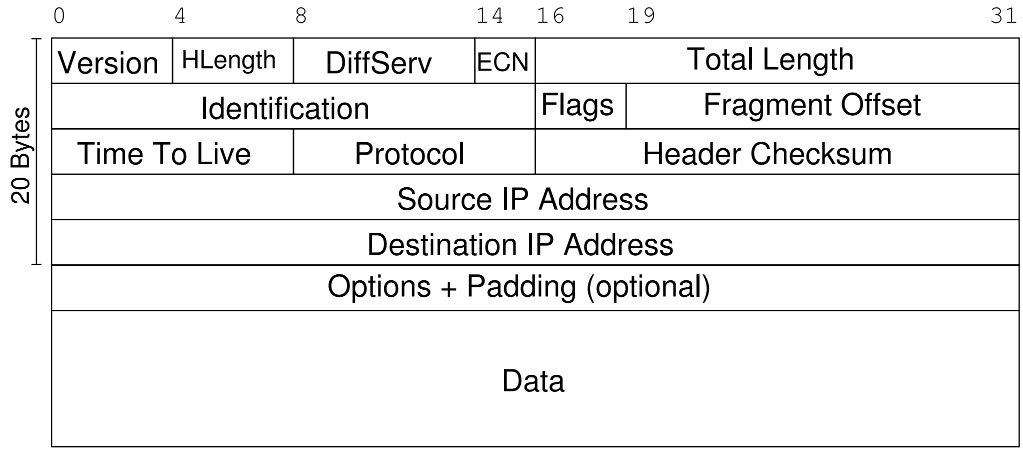


Figure 1: IP Datagram Format. Flags: Reserved, Don't Fragment, More Fragments

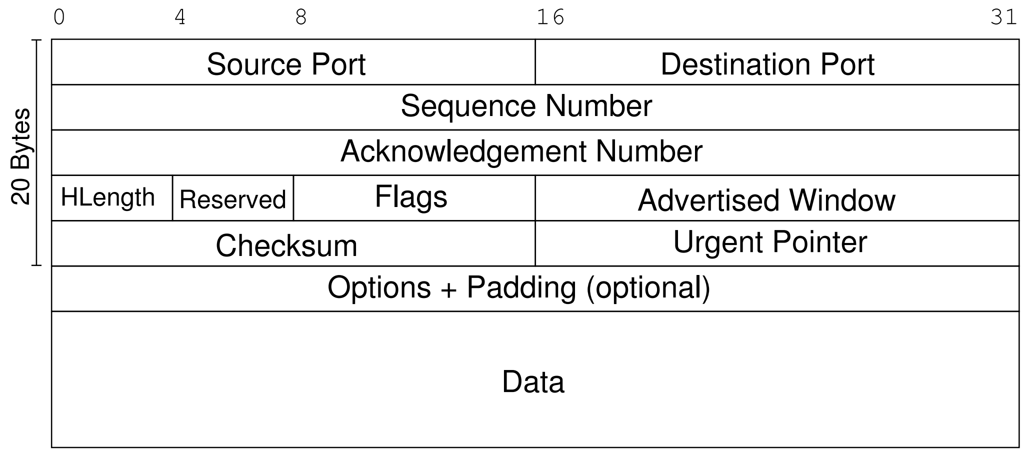


Figure 2: TCP Segment Format. Flags: CWR, ECE, URG, ACK, PSH, RST, SYN, FIN

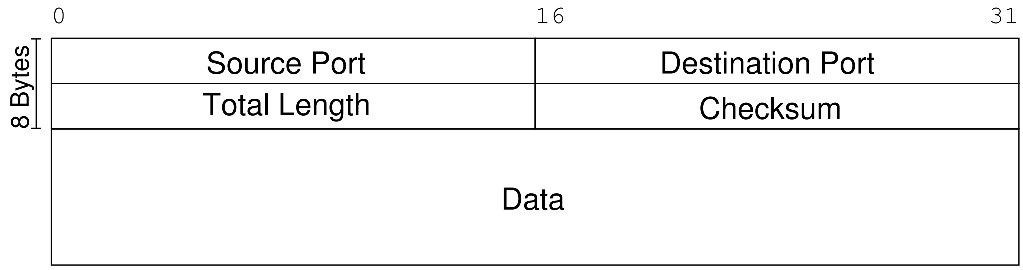


Figure 3: UDP Datagram Format

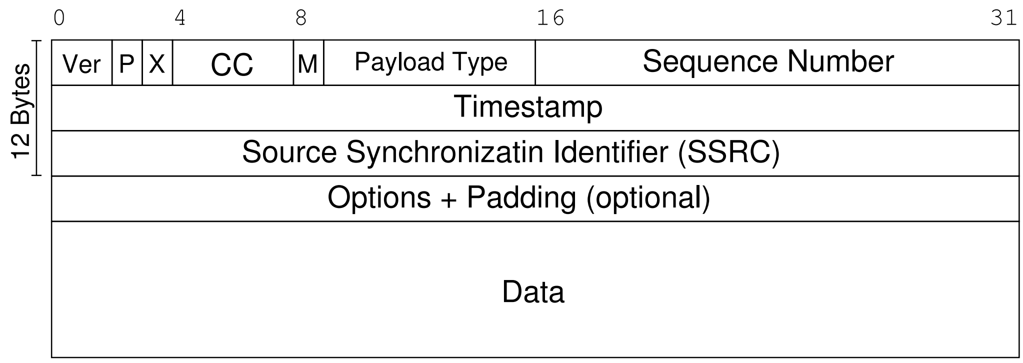


Figure 4: RTP Packet Format. P: Padding; X: Extension; CC: CSRC count; M: Marker