TSN: Lecture 16
Connection Routing

Topics Covered

- How is Connection Routed?
- DC Pulsing Signals
- Single Frequency (SF) Signaling
- Multi-Frequency (MF) Digit Signals
- Automatic Routing of Subscriber Dialed Calls
- Common Channel Signaling
- Signaling System 7
- Main SS7 Message Types
- Local Number Portability

How is Connection Routed?

- A connection is directed from an originating line or trunk to a destination line or trunk outlet by setting the correct numbers in the connection memory associated with the switching matrix.
  - The choice of destination or outlet is based on an internal translation table which uses a part of the directory number as input (e.g. NPA or NXX) and physical equipment identification number (trunk number or ILAN) as output.
  - The contents of this translation table may be changed from time to time (or another table substituted) to accommodate:
    - Malfunction or traffic saturation of certain links (use of alternate path or route)
    - Known or expected better choices for economic or traffic reasons
      - Time of day changes in traffic in various time zones
      - Utilize different price strategies on leased or outgoing trunks

Historical North American* Signaling Methods

- DC (direct currrent) or baseband dial-pulse signaling 1900-1940s
- SF (single frequency) impulsive tone signaling for supervision and dialing. Needed for FDM multiplexing.
- MF (multi-frequency) dialing digit representations.
  - The above 3 methods are each historically called "in band" even when analog frequency band is not involved (for example, in T-1).
- When digital multiplexing (eg, T-1) arrived, MF dialing signals and "robbed bit" supervision signaling was used.
- Many telephone networks formerly or presently use(d) in-band tone signals to represent dialed digits (DTMF "touch tone," or another set of audible tones called multi-frequency MF)
  - DTMF uses two simultaneous tones taken from a set of 8
  - MF uses two simultaneous tones taken from a different set of 5
- Common channel signaling (SS7 ISUP) is dominant today
  - Earlier versions were SS6 and SS7 TUP.

*R2 and other signaling methods used outside of North America are not covered in this lecture.

DC Pulsing Signals

- Before dial telephones, for inter-office (inter-switch) signaling (usually in same city) the human "operator" connected through a trunk to another human operator at destination switch. Requests for a destination number were verbal. When subscribers ended the call, buzzers at each cord board signaled the operators to manually "tear down" (unplug) each link.
- Earliest step switch dial signaling methods accessed specific inter-office trunks in response to dialing the first (typically) 3 digits. The remaining dialed digits were out-pulsed in the same form as "locally" dialed signals (that is, brief 40 ms interruptions in the loop current) on the the same wires as the eventual voice channel, and operated the last stages of step switches in the destination switch. A "long" interruption in the loop current ultimately caused automatic disconnect of all the step switches involved in the call, just as for a "local" (same switch) call.
- This system was difficult to evolve further:
  - Required many permanently installed, directly-connected but not always frequently used inter-office trunks.
  - Long distance voice transmission with electronic (vacuum tube) amplifiers began about 1914. (Before that, very costly thick low-resistance trunk lines were installed between some cities.) Amplifiers also required conversion from 2-wire switching (used in each end switch) to 4-wire switching in the trunk link.

Single Frequency (SF) Signaling

- In the 1920s and 1930s, analog frequency division multiplexing (FDM) was introduced into the long distance telephone network. Typically 12 conversations were multiplexed on the same 4 trunk wires, by means of amplitude modulation of each conversation (in each direction) onto a distinct carrier frequency.
  - Similar to having 12 radio broadcasting stations, with 12 pre-tuned radio receivers. Except the radio frequency signals were all carried by wires and *not* "over the air."
  - DC or baseband pulsing was not feasible with FDM.
- In each conversation signal in each direction, a single frequency (2600 Hz) tone was transmitted in the voice channel to indicate the *absence* of subscriber loop current.
  - Steady SF tone indicated that the channel was idle, or that the subscriber at the relevant end had hung up.
  - Out-pulsing of the SF tone at the beginning of a call setup conveyed dialing digits as from a rotary dial.

Multi-Frequency (MF) Digit Signals

- SF for dialing was "slow," and was also susceptible to theft of service fraud from SF tones generated at the originating subscriber end by "hackers."*
- MF dialing encoded each digit, and also a "start" (KP) and "stop" signal, each by means of a pair of tones taken from a menu of five distinct audio frequencies.
  - MF dialed digit signals were initially used with SF supervision.
  - This required a more sophisticated multi-tone receiver to decode the tones and produce electric current pulses compatible with the switching equipment.
  - New types of switch equipment (example: crossbar) were introduced in the 1930s that did not need sequential digit pulsing and thus connected the call faster when MF was used.
  - MF signaling can out-pulse 10 digits in one second. Baseband and SF out-pulsing could take up to 10 seconds for this.

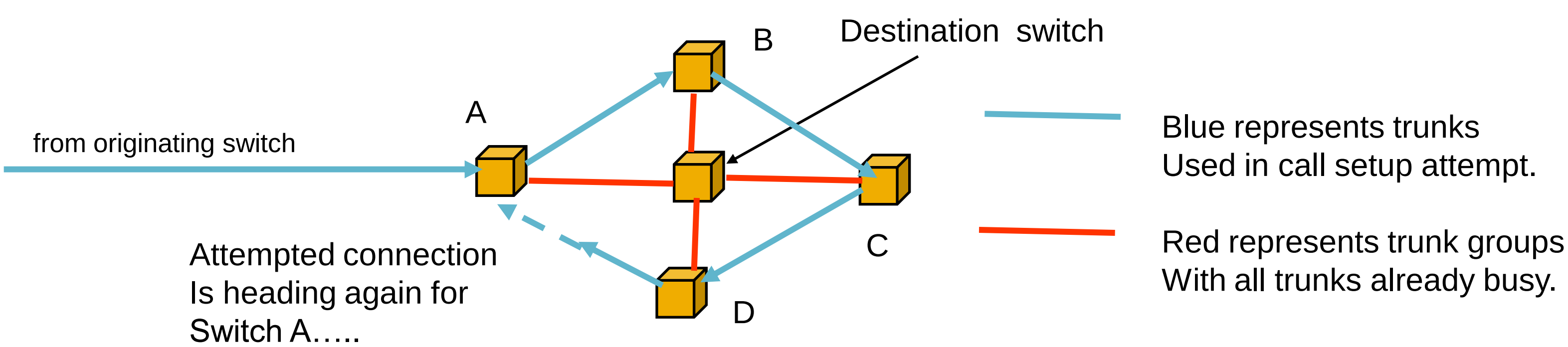
*Large scale "hacker" fraud did not become a widespread problem until the 1960s

MF with Digital Multiplexing

- When T-1 transmission was introduced (1961), MF was still used to convey dialed digits.
  - Supervision was signaled via “robbed bit” signaling (described in EETS8302 lecture notes and in Bellamy recommended reference)
- The same advantages and disadvantages still applied to MF encoding of the destination number
- Plus a new requirement:
  - Not flexible for evolving new services. Next slide shows and example where other information is needed.

Automatic Routing of Subscriber Dialed Calls

- Automatic routing of long distance calls via several intermediate switches (first introduced in late 1950s) called for more sophisticated signaling
- Knowing only the destination telephone number was not sufficient
  - A problem called "ring around" or "looping" can occur when an automatic call routing system tries several suburban intermediate switches surrounding a destination city, and eventually revisits the same intermediate switches
  - The call signaling must identify each call so that the previous intermediate switches on its attempted route are not visited twice…
    - Better to signal to originator to "try again later" and abandon the call
    - Unlimited "ring around" will eventually gobble up *all* circumferential trunk channels, making the situation worse than it already is.



Common Channel Signaling

- The first common channel signaling system (1970s, CCITT No.6 or common channel interoffice signaling – CCIS) used fixed length fixed structure messages via a 2400 bit/second modem in a pre-designated channel.
- Signals in the "common"channel controlled call setup and disconnect and some routing features in any voice channel.
  - Prevented hacker fraud
  - Still not versatile for future evolution
- Replaced starting in 1980s with common channel system number 7.
  - Higher bit rate, faster call setup
  - Versatile for future evolution

Signaling System 7`

- Almost a world wide standard
  - Some variations in versions and features in different national telephone networks
- High bit rate. Two implementations:
  - Uses one voice channel, carrying 56 kb/s or 64 kb/s
  - Alternative implementation uses all of a T-1 bit rate (1536 kb/s) except the synchronizing bit.
- Flexible
  - Message formats provide for future addition of new parameters
- Reliable
  - Typical SS7 network has extensive duplication of data packet switches and geographically diverse alternate transmission channels, and includes continual built-in channel testing and route around failed links/switches.

Main SS7 Message Types

- Standard SS7 uses 4 Message types for ordinary call setup, but other types exist and more can be added
  - Initial Address Message (IAM) orig->dest
    - Contains dialed and originator's numbers, and other information
  - Address complete msg. (ACM) dest->orig
    - Indicates if destination Line is ringing or busy
  - Answer (ANS) dest->orig
    - Indicates that destination line has been answered.
    - SS7 is designed to allow postponing assignment of a voice channel until this event. This feature uses voice channel capacity much more effectively but will be implemented only in the future when the world PSTN is 100% SS7.
  - Release (RLS) and its acknowledgements- either direction
    - Causes disconnect process.
- In addition to the "standard" parameters of each message type, new parameters can be added when required.

SS7 Services Examples:

- Calling Line Identification (Caller ID)
  - Originator telephone number sent as a parameter in the IAM to the destination switch. Sent to destination line between first two rings via a modem tone, if not “blocked.” This was the economic “killer app” for SS7
- Call back to last unanswered caller
- Call completion to busy subscriber
  - Monitors busy line when denied caller requests, and establishes connection with caller after it hangs up.
- Redirect “800” calls and other pre-designated calls to alternate numbers
  - When you dial 1 800 DOM-INOS, directs call to the nearest Domino’s pizza retail store. Uses data base indexed by first 6 digits of the originator’s number.

Future Telephone Signaling

- Voice over Internet Protocol (VoIP) using speech digitally coded at low bit rate into data packets has caused great interest in use of the Internet and the traditional PSTN to set up calls. Two protocols are in use for these applications:
- H.323, originally designed for multi-media and conference calling via Internet, is the first historical method. Similar in many ways and designed to be backward compatible with SS7. Uses binary number parameters in messages.
- Session Initiation Protocol (SIP) is simpler for non-telephone experts to understand, and uses alphabetic (ASCII) characters and printable numeric digit parameters.

Advanced "Intelligent" Network (AIN)

- Numerous advanced features utilize SS7 messages. Examples:
  - Call completion to busy subscribers (CCBS). Originator dials *66 (auto redial‡) upon reaching a busy line, then hangs up.
    - Destination switch automatically sends a special SS7 message when desired but busy destination person hangs up.
    - Origination switch rings back the originating/requesting line. If origination person answers, origination switch sends a message to destination switch causing destination line to ring.
    - If destination person answers, conversation proceeds.
- SCP data base can "translate" 800/888/877 dialed numbers into destination numbers appropriate to the caller, the time of day, etc.
  - Callers to Sears Roebuck, Domino's Pizza, actually reach the geographically closest retail outlet, or the east coast customer order center in the morning and the west coast center later in the day, so order takers work shorter hours at each such location!
  - Translation based on origination calling line ID and extensive data base relating each NPA/NXX to the directory number (DN) of the nearest retail outlet, etc.

‡ This marketing name is misleading, because the network does not actually re-dial the desired destination repeatedly. It is "dialed" only once, after the originator answers the ring back.

Local Number Portability

- The objective of LNP is to promote local service competition.
- The FCC (and some other national telecom regulators such as OfTel in UK) have legally mandated "portability" of telephone directory numbers when a subscriber changes from an existing ("donor") LEC switch to a new ("recipient") competitive local exchange carrier (CLEC) telephone service provider switch.
  - In North America, several centralized duplicated SCP/STP data bases provide a translation between the subscriber's DN and a special "dummy" number (called a location routing number -LRN) located on the new recipient CLEC switch.
  - After this translation, the call is routed to the recipient destination switch using the LRN's NPA/NXX. The *actual* dialed number is carried in a separate special information parameter in the IAM message.
- When the call is routed to the recipient destination switch, the destination switch connects the call to the internal subscriber line designated in a *special* translation table which relates such "ported" DNs to the physical lines (ILANs) in the switch used for that purpose.

Proposed "System Beta"

- Your instructor proposed a software-based network technology* in 1998 to allow *all* of one subscriber's business or residence telephone lines to each be dialed using the *same* 7 (or 10) digit decimal telephone number.
  - The subscriber may group all business lines (voice, fax, cell/PCS, pager, data, etc.) under one number and all residential lines (teenager's line, home fax, etc.) under another number
  - The different lines in a group are distinguished from each other by means of functional purpose (FP) codes which are mostly pre-set by the user
    - FP codes for each line are stored in a suitable data base (perhaps the same data base is used for LNP!)
    - The internal network SS7 signal messages carry the FP codes in separate parameters of existing messages
  - Multi-use lines have a preset normal FP (typically voice) which can be temporarily superceded on individual calls by a dialed prefix (e.g., a fax machine automatically dials a *329 prefix, indicating that the desired destination is a fax line *during this call only*).

*US patent 6,076,121 issued June 13, 2000, and foreign patents pending.

Beta Status

- Aside from making it easier to remember all the different telephone numbers of your correspondents, System Beta reduces the problem of telephone number exhaustion by reducing the quantity of telephone *numbers* for most subscribers to just one or two, regardless of the quantity of lines in service!
  - In some cases *most* of the lines used by a subscriber will have internal non-decimal BCD digits in their *internal* network telephone numbers. This is not visible to the end user, but it stops the present growth in use of multiple decimal telephone numbers by subscribers.
- Also permits blocking undesired callers by functional purpose (e.g., block all unsolicited sales calls) and other new optional features
  - FCC now considering System Beta due to above benefit and uses for the disabled (e.g. automatic routing of special 911 or teletypewriter for deaf calls)
- Estimated complete *one-time* development and installation cost is approximately $7 Billion to the telephone industry (mostly administrative and testing costs).
  - At present the telephone industry spends about $1billion each year due to area code changes, with no end in sight to this recurring cost. Possibly additional $7 Billion in 2005 to 2020* to change to 4-digit area codes.
  - Cost of all area code changes and 10-digit dialing to the public is estimated to be much more ($50 Billion to $150 Billion).
- System Beta was tabled by the T1S1.3 standards committee which defines North American SS7 signaling message standards.
  - Waiting for direction from a major carrier or FCC to take it off the table.
- Could permit "recombining" previously split area codes, restoring 7-digit local calling, in perhaps 2 to 8 years after installation.

*Industry predictions of date when 11 digit NANP will be needed varies with the source and the date of estimate.