Designing a Voice over IP Network

Introduction

- The design of any network involves striking a balance between three requirements.
  - Meeting the capacity needed to handle the projected demand (capacity)
  - Minimizing the capital and operational cost of the network (cost)
  - Ensuring high network reliability and availability (quality)
- Meeting one or more of the requirements often means making sacrifices elsewhere.
- What is the acceptable degree?

An Overall Approach

- Understanding the expected traffic demand
  - Where traffic will come from and go to
  - What typical per-subscriber usage is expected
- Establishing network design criteria
  - Build-ahead, voice-coding schemes, network technology (such as softswitch versus H.323)
- Vendor and product selection
- Network topology, connectivity and bandwidth requirements
- Physical connectivity

Design Criteria [1/2]

- Build-Ahead or Capacity Buffer
  - Avoiding the necessity for constant redesigning as traffic demand increases
  - Providing a buffer in case traffic demand increases faster than expected
- Fundamental Technology Assumptions
  - H.323 vs. Softswitch
  - MGCP vs. MEGACO
  - Should we use external SGs with Sigtran or deploy MGCs that support SS7 directly?
- Network-Level Redundancy
  - E.g., Failure of MGCs, Failure of network interfaces

Design Criteria [2/2]

- Voice Coder/Decoder (Codec) Selection Issues
  - Actual coder/decoder to use
  - Packetization interval
  - Silence suppression
- Blocking Probability
  - A call will be blocked due to a lack of available channels.
  - The Erlang is the standard measure of traffic on a circuit-switched network.
    - One Erlang corresponds to a channel being occupied for one hour.
  - Depending on the number of available channels and the amount of offered traffic, there is a statistical probability that a channel will be available when a user wants to make a call.
- QoS Protocol Considerations and Layer 2 Protocol Choices (e.g., Frame Relay, ATM or PPP)

Product and Vendor Selection

- Generic VoIP Product Requirements
  - Node-Level Redundancy
    - N+1 redundancy
  - Node Availability
    - 99.999 percent availability
    - Mean Time Between Failure (MTBF) values provided by vendors for each component of a given node
  - Alarms and Statistics
    - For the network operator to fully understand the performance of the network
- Element Management
  - E.g., SNMP for interfaces between the network elements and EMS

Traffic Forecasts

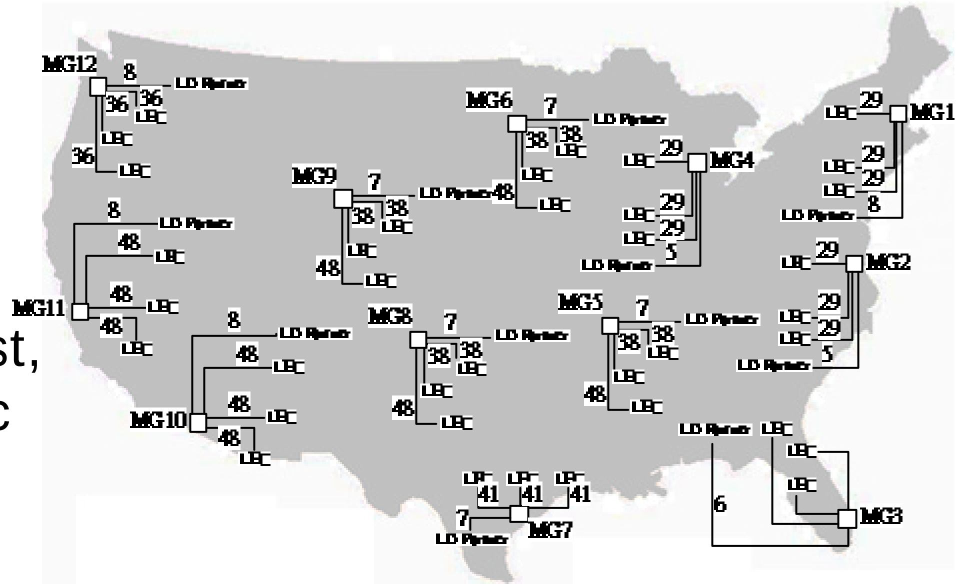
- Voice Usage Forecast
  - (MoUs per subscriber per month) x (fraction during work days) x (percentage in busy hour) / (work days per month)
    - E.g., 120x0.6x0.2/21=0.686 MoU/sub/busy hour
    - 0.686/60=0.0114 Erlangs/sub/busy hour
    - The driving factor for the network elements that reside in the bearer path
  - Busy-hour call attempt (BHCA)
    - Assume that the average call length is 5 minutes (300 seconds).
    - =Erlangs/MHT (average call length) =0.0114x3600/300=0.137
    - The critical factor for call-control entities such as MGCs
    - A subscriber with 120 MoUs per month will make 0.137 calls each busy hour.
- Traffic Distribution Forecast

Network Topology

- How many network elements of a given type will be in each location
- The bandwidth requirements between those network elements and the outside world

MG Locations and PSTN Trunk Dimensioning

- At least 1 MG in each of the 12 cities where the service is provided
- To determine the size of the trunk groups to the PSTN
  - From Voice Usage Forecast, we know how much traffic we will send.
  - From Traffic Distribution Forecast, we know how much traffic we will receive.



MGC Quantities and Placement

- Assume that BHCA is the limiting factor.
- A call passes between two MGs controlled
  - By the same MGC
  - By different MGCs
- Determining the number and location of MGCs can be an iterative process.
  1. An initial estimate of the number of MGCs
  2. To allocate MGs to MGCs
  3. To determine the total BHCA to be supported by each MGC
  4. See if the initial MGC allocation fits within the MGC BHCA limit.
  5. If not, go to 1.

Calculating VoIP Bandwidth Requirements

- The bandwidth required between MGs for VoIP traffic
- The bandwidth required for a single call depends on the following factors.
  - Voice-coding scheme
  - Packetization interval
  - The use of silence suppression
  - Probability of excessive packet collision
    - Packet will be lost or delayed as a result of too many speakers talking at one time.

Peak in the Number of Simultaneous Speakers

- Consider n speakers. If voice activity is 40 percent, then the probability of an individual user speaking at a given instant is 40 percent.
- The probability that exactly x subscribers are speaking at a given time
  - $Pa(x) = (n,x)\ p^x(1-p)^{n-x}$, where $p=0.4$
- The probability that there are no more than x speakers at a time
  - $Pb(x) = Pa(0)+Pa(1)+\ldots+Pa(x)$
- To determine the value of x
  - Seeking $Pb(x)=0.999$ or greater
- Normal distribution function instead of binomial distribution due to computation complexity

Bandwidth Requirement

- VoIP Bandwidth
  - Voice packet size + 40 octets (for IP, UDP and RTP) + WAN layer 2 overhead + MPLS overhead (if applicable)
  - RTCP bandwidth should be limited to about 5% of the actual VoIP bandwidth.
- Signaling and OA&M Bandwidth
  - Between MGC and MG
  - Between MGC and SG
  - Between SG and STP
  - Between MGC and MGC
  - Between each network element and EMS

Physical Connectivity

- To determine how we will connect the different cities to provide the bandwidth we need
- Each city has an alternative path to every other city to ensure the network does not fail.