1. Introduction

- Capacity planning is primarily concerned with understanding & quantifying
  - Application behaviour
  - User behaviour
  - Traffic characteristics
  - Network performance characteristics, such as network utilization

- It lays the foundation of network design!
- Advanced protocols, dynamic traffic patterns and characteristics, and peer-to-peer internetworking has changed capacity planning into more of a heuristic guesswork approach than one based on calculation.
- The traffic matrix is no longer a two-dimensional spreadsheet, but a multidimensional matrix including variables such as protocol types, multiple protocols, multiple traffic-flow patterns, multiple technologies, circuit options, and more.

Requirements Definition

Outline

1. Introduction
2. Throughput Calculation
3. Traffic Engineering Basics – Traffic Characteristics and Source models
4. Traditional Traffic Engineering – Voice Traffic Modelling
5. Queued Data and Packet-Switched Traffic Modelling
6. Designing for Peaks
7. Delay
8. Availability and Reliability
9. Reaction to Extreme Situations
10. Network Performance Modelling
11. Creating Traffic Matrix
12. Capacity Planning and Network Vision
1. Introduction

- Capacity planning procedures:
  - **Step 3:** Baseline the existing network
    Take regular baseline measurements to capture traffic profile for future use, including:
    - Bandwidth utilization in the cases of broadcast/unitcast
    - Bandwidth utilization of protocols
    - Packet/frame size distribution
    - Background error rate
    - Collision rate
  - **Step 4:** Making traffic projections
    - By hand: using performance estimation techniques
    - By commercial analytical tools
    - By discrete event simulation tools, which can help to form a more detailed view of network utilization and the network's impact on the performance of applications.

2. Throughput Calculations

- **Throughput** is the actual amount of user protocol data that is transmitted over the access circuit and received by the network node. Throughput can also be measured end-to-end across the entire network.
- Access-circuit speeds are represented as **total capacity**. The actual throughput that the user receives can be much less than that, depending on the protocols and equipment used.
- **Protocol overhead** includes: header and trailer data wrapped around the user data, time spent waiting for acknowledgements when data is not be transmitted.
- **Throughput = Total Capacity - Protocol Overhead**
- Users are concerned with throughput, not the capacity.

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- **Packets, Frames, and Cells per Second**
  - If one were to ignore the intricacies of each higher-layer protocol, the maximum achievable throughput could be calculated.
    - **Example:** on a 56 kbps capacity line, an X.25 packet switch using 128-byte packets could pass up to 55 Packets Per Sec.

- **Effects of Overhead**
  - The above PPS calculation does not account for the overhead incurred in the switching and protocol handshaking operations.
    - **Example:** take a 512 kbps frame relay fractional T1 access line. The frame size of 1024 bytes with overhead of 13 bytes per frame is used. The actual Frame Per Sec throughput is:
Traffic Engineering and Capacity Planning

Effects of Overhead

Example: now change the frame size to 56 bytes. The overhead is still 13 bytes per frame. So we have

\[
512 \text{ kbps} \times \frac{1}{8} \text{ Byte} \times \frac{1}{69} \text{ Bytes} \approx 928 \text{ FPS}
\]

Throughput degrades dramatically!

\[
\text{Overhead 1} = \frac{13}{1037} = 1.25\% \\
\text{Overhead 2} = \frac{13}{69} = 18.84\%
\]

- The larger frame sizes are more efficient and provide higher line throughput than the smaller ones, but up to a certain point.
- In packet switching, the larger the packet size, the higher the probability of error, causing data to require retransmission.
- For noise lines, throughput can be increased by decreasing packet size. The added overhead is offset by reduced retransmissions.

Traffic Engineering and Capacity Planning

Traffic Engineering and Capacity Planning

3. Traffic Engineering Basics – Traffic Characteristics and Sources Models

- Source Model Traffic Parameter Characteristics
  - Deterministic parameters are based upon a specific traffic contract, with conformance verifiable on a unit-by-unit basis.
  - The agreement as to the traffic throughput that achieves a given performance is unambiguously stated.
  - The probabilistic (also called stochastic) model is typically measurable only over a very long-term average.
  - Since the method and interval for computing the average can differ, conformance testing defines the details of the measurement method.
  - Specification of the statistical model is also required.

Traffic Engineering and Capacity Planning

General Source Model Parameters

- Burstiness is a measure of how infrequently a source sends traffic. A source that infrequently sends traffic is very bursty.
  
  \[ \text{Burstiness} = \frac{\text{Peak Rate}}{\text{Average Rate}} \]

- Source activity probability is a measure of how frequently the source sends, defined by the probability that a source is bursting.
  
  \[ \text{Source Activity Probability} = \frac{1}{\text{Burstiness}} \]

- Utilization is a measure of the fraction of a transmission link's capacity that is used by a source during a time interval.
  
  \[ \text{Peak Utilization} = \frac{\text{Peak Rate}}{\text{Link Rate}} \]

- All protocol and switching overheads should be accounted for in the calculation of utilization.
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4. Traditional Traffic Engineering

- **Statistical Behavior of User Traffic**
  - User information arrives at the network node based on statistical arrival rates. Therefore, statistical approximations can be used to model these traffic patterns.

- **Voice Traffic Modeling (Erlang Analysis)**
  - Erlang is the measure used in analog voice communication systems for estimating user demand. It is defined as
    \[
    \text{Erlang } E = \lambda \cdot \tau = \sum_{n=1}^{\infty} \lambda^n \cdot \tau_n
    \]
  - where \( \lambda \) is the call arrival rate in calls/hour, and \( \tau \) is the average call holding time in hours.
  - Example: If 100 calls of 150 second duration, 200 calls of 100 second duration, and 300 calls of 50 second duration within one hour, the number of erlangs would be 13.89.

Traffic Engineering and Capacity Planning

5. Queued Data and Packet-Switched Traffic Modeling

- While Erlangs work well predicting voice network and circuit-switched traffic rates, they do not work well with packet-switched networks.
  - In packet-switched networks, some level of queuing is employed so that packets are queued in buffers and transmitted when congestion ceases, rather than being immediately blocked.
  - Packet-switched networks provide a mix of protocol and traffic types, whereas voice and circuit-switched networks provide point-to-point, transparent homogeneous transport of information.

  Therefore, packet switching demands a different analysis of traffic handling.

- **Queueing System Models Notation**
  - A/B/s/w/p
    - A: arrival process of new calls/packets (A=M, G or D)
    - B: departure process of served calls/packets (B=M, G or D)
    - M: Markovian; G: General; D: Deterministic.
    - s: number of queue servers (s>0)
    - w: size of waiting room (or number of buffer positions, w>0)
    - p: source population (or number of users, p>0)

- **Queued Data and Packet-Switched Traffic Modeling**
  - Erlang-B: M/G/s/s ➔ voice blocked calls cleared and FDM/TDM.
  - Erlang-C: M/G/s/k ➔ voice blocked calls held or operator services (this model is used when k>s).
  - Packet: M/G/1 ➔ packet, frame, and cell networks (assume infinite waiting room and population).
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Markovian Queueing Systems Models
- M/D/1 model
  - Constant length bursts \( \not \) accurate
  - Buffer unit is packet, frame, or cell \( \not \) accurate
  - Difficult to analyze
- M/M/1 model
  - Random length bursts with a negative exponential distribution (memoryless, Markov) \( \not \) accurate
  - Buffer unit is burst \( \not \) accurate
  - Simple to analyze

Utilization and Capacity Calculations
- Utilization (or offered load) is a unitless measure to represent the average fraction of the resource capacity that is in use.
  \[ \rho = \frac{\lambda}{\mu} \]
  - \( \lambda \) is the average number of arriving bursts per second;
  - \( \mu \) is the average service time per burst.
- The probability that there are \( n \) bursts in the M/M/1 system is given by
  \[ \Pr(n \text{ bursts in M/M/1 system}) = \rho^n(1 - \rho) \]
- The average queue size is
  \[ N - \rho = \frac{\rho}{1 - \rho} \]
- The average queueing delay (waiting time) is
  \[ w = \frac{\rho}{\mu(1 - \rho)} \]
- The average delay is equal to the sum of the waiting time (in the queue) and the service time (at the server), specifically
  \[ d_{avg} = w + \frac{1}{\mu} \]

Application of M/D/1 and M/M/1 Queueing System with Cells

Relationship Between Utilization and Queue Size
Traffic Engineering and Capacity Planning

Markovian Queueing Packet Switching System Example

- If a packet switch has 5 users, each transmitting 10 messages per second at 1024 bits per message, with the packet switch operating over a 56 kbps trunk, the following applies:

\[ \lambda = 5 \times 10 = 50 \text{ mps; } \mu = \frac{56000 \text{ bps}}{1024 \text{ bps}} = 54.6875 \text{ mps} \]

\[ \rho = \frac{\lambda}{\mu} = 0.914 \Rightarrow \text{Utilization;} \ N = \frac{\rho}{1 - \rho} = 10.63 \text{ messages in system} \]

\[ N - \rho = 9.716 \text{ messages in queue} \]

\[ w = \frac{N}{\mu} = 0.194 \text{ seconds} \Rightarrow \text{average waiting time} \]

\[ d_{avg} = w + \frac{1}{\mu} = 0.194 + 0.0183 = 0.212 \text{ seconds} \Rightarrow \text{average delay} \]

Traffic Engineering Complexities

- Realistic source and switch traffic models are not currently amenable to direct analysis, only approximations can be derived under certain circumstances.

- Simulations are time consuming and, in the case of modeling complex high-speed technologies like ATM, cannot effectively model low cell-loss rates since an inordinate number of cells must be simulated.

- Constantly changing source, switch and network characteristics create a moving target for such traffic engineering models.

Buffer Overflow and Performance

- The probability that there are \( n \) packets in the M/M/1 system is

\[ P_n = \text{Pr}\{ \text{n packets in M/M/1 system} \} = \rho^n \cdot (1 - \rho) \]

- The probability of overflowing a buffer of size \( B \) packets is

\[ \text{Pr}\{n > B\} = \sum_{n=B+1}^{\infty} P_n = \rho^{B+1} \]

Cell Buffer Overflow Analysis

- The overflow probability for a buffer of size \( B \) cells (M/M/1/B) is approximately the probability that there are more than \( B/P \) bursts in the infinite queue system (M/M/1), so we have

\[ \text{CLR} = \text{Pr}\{ \text{Cell Buffer Overflow} \} \approx \rho^{\frac{B}{P}} \]

where CLR stands for Cell Loss Ratio and \( P \) represents the average number of cells in contained a PDU burst.

- Buffer overflow probability in frame and cell networks increases as the higher layer PDU sizes (\( P \) value) increase.

- Therefore, the required buffer size for achieving an objective CLR is approximated by

\[ B = P \cdot \log_p \text{CLR} \]

- The shared output buffer scheme has a marked improvement on buffer overflow performance because of sharing a single, larger buffer among many ports, and it is unlikely that all ports are congested at the same time.
Traffic Engineering and Capacity Planning

Switch Buffering Performance

Overflow Probability versus PDU Burst Size

Shared versus Dedicated Buffer Performance

Shared versus Dedicated Buffer Performance

Traffic Engineering and Capacity Planning

Switch Buffering Performance

Overflow Probability versus PDU Burst Size

Shared versus Dedicated Buffer Performance

Shared versus Dedicated Buffer Performance

Traffic Engineering and Capacity Planning

Switch Buffering Performance

Overflow Probability versus PDU Burst Size

Shared versus Dedicated Buffer Performance

Traffic Engineering and Capacity Planning

Shared versus Dedicated Buffer Performance

Statistical Multiplexing Gain

- Statistical multiplexing attempts to exploit the on/off, bursty nature of many source types.
- As more and more sources are multiplexed, the statistics of this composite sum become increasingly more predictable.
- Statistical Multiplex Gain

\[ G = \frac{N}{N + 4b/\eta - b} \]

- The rate of any individual source should be low with respect to the link rate \( \eta \), and the burstiness of the sources \( b \) must be high in order to achieve a high statistical multiplex gain \( G \).
LAN traffic modeling is a difficult process and is subject to further study. Many LAN traffic characteristics, such as performance and throughput based on number of users, have been thoroughly studied and provided by equipment vendors.

The Token Ring throughput increases when the number of users increases because less time is spent token passing, whereas on the Ethernet (CSMA/CD) LAN the throughput decreases as the number of users increases due to the increased likelihood (and number) of collisions.

LAN bridge designs are concerned primarily with frames forwarded per second and frames filtered per second. Packet and frame forwarding and filtering buffers, as well as the depth of LAN address table memory, should also be considered in the designs.
**Traffic Engineering and Capacity Planning**

LAN Throughput Comparison

The DQDB (Distribute Queue Double Bus) MAN Traffic Modeling

- The DQDB bus operates as a LAN, but handles calls similarly to the Erlang method, where messages contending for the bus have to wait until they can reserve a space on the bus.
- The required capacity of a DQDB MAN to handle all user traffic is calculated with the sum of the $\lambda$'s (packets/sec) of the local, remote traffic from and to the MAN, and the pass-through traffic, i.e.

  $\sum \lambda_l + \sum \lambda_{RF} + \sum \lambda_{RT} = \lambda^*$

  where all $\lambda$'s are the sum of the users in that category and $\lambda^*$ represents the minimum required capacity of the local MAN.
- Since MANs often provide high-bandwidth connectivity to a small number of users, the traffic approximations just discussed become valid (where aggregations tend to have Poisson distributions). Huge bursts on the MAN can dwarf the normal large packet transmissions normally seen on the LAN.

**Characteristics of LANs Attached to MANs**

- The probability that a LAN time slot will be busy is given by

  $\rho_B = \frac{\text{Avg. number of slots consumed by burst}}{\text{Time of burst}} = \frac{\lambda}{\mu}$

- The probability that a LAN will transmit onto a particular MAN slot is $\rho_m = \rho_B \cdot \rho_{\text{Inter-LAN}}$ ($\rho_{\text{Inter-LAN}}$ the fraction of inter-LAN bursts).
- The average utilization (or throughput) of the DQDB MAN can be approximated by $\mu_B = N \cdot \rho_m / y$ (assume there are $N$ LANs connected to MAN and the MAN is $y$ times faster than each LAN).
- The average M/M/1 delay is proportional to $1/(1 - \rho_B)$.

- **Queueing power** is defined as the ratio of throughput to delay, i.e.

  $P = \frac{\text{Throughput}}{\text{Delay}} = \frac{\mu_B (1 - \rho_B)}{\rho_B} = \frac{N \cdot \rho_B}{y} \left(1 - \frac{N \cdot \rho_B}{y}\right)^2$

- The optimum number of LANs that can be connected to the MAN can be solved as

  $\frac{\partial P}{\partial N} = 0 \Rightarrow N_{\text{opt}} = \frac{y}{2 \cdot \rho_B}$
6. Designing for Peaks

Any network must be designed to handle the busiest traffic periods, e.g. the Mother’s Day is the busiest calling day of the year.

- **Standard Busy Hour Calculations**
  - When call arrival rate ($\lambda$) is calculated and applied to the formulas for traffic load, the designer will use a value measured or estimated during the peak busy period, which is called the "busy hour".

- **Data Equivalent of Busy Hour**
  - Data networks have their own equivalent of busy hour, which is the time period when the highest throughput is required.
  - Cost tradeoff between network scale and transmission delay.
  - Since its burstiness, data traffic is more accurately analyzed with multiple snapshots down to a “busy minute” basis.

7. Delay

- When a network begins to slow down because of buffering, retransmissions, and/or any other time-affecting factor, its users will begin to experience delay and may experience loss of traffic.
- Delay and loss will cause response time and throughput to degrade, application time-outs and retransmissions to occur. They are the major considerations in application designs.
- The accumulation of delay variation in multi-hop networks is important to delay-variation-sensitive applications, such as video and audio.
- Appreciable variations in delay (or jitter), on the order of 50 ms or more will be observed by most users.

### Causes of Delay

- Propagation path length
- Line speed
- Number of access hops
- Hardware and software interface buffers
- Load on every component across the path
- Hardware/processor elements traversed (each adds delay)
- Window sizes
- Bit-setting selections
- Memory and buffers
- Pad functions
- Address database look-up
- Address verification
- Changes in traffic load
- Filtering, forwarding, and processing packets, frames, and cells

### Circuit-, Message-, Packet-, and Cell-Switching Delay Basics

- In **circuit-switching**, the delay incurred by a user is based upon the message size and the time it takes to transmit it over the available bandwidth.

  **Example:** if a 2 Mbit file is transmitted over a 128 kbps frame relay trunk, the total delay in transmission (ignore network and protocol delay) is $2 \times \frac{15.625}{128} = 2.34375 \text{ sec.}$

- In **message-switching**, delay closely resembles that found in frame relay networks. The total delay is calculated the same as in circuit switching, but multiplied by the number of nodes the message must traverse, minus 1.

  **Example:** if the 2 Mbit file is transmitted through a four-node frame relay network (not counting the origination and destination nodes), with all nodes passing data at 128 kbps.
Traffic Engineering and Capacity Planning

Assume the user has the entire bandwidth of each link, the total delay is given by

$$\frac{2 \text{ Mbit}}{128 \text{ kbps}} (4 + 2 - 1) = 78.125 \text{ sec.}$$

In packet-switching, the total message delay is calculated as

$$\frac{p}{c} \cdot (n+1) + \frac{m-p}{c}$$

where p: packet size; c: transmission rate of the medium; n: number of intermediate nodes; m: message size.

**Example**: if the 2 Mbit file is transmitted over the four-node (not counting the origination and destination nodes) network with 56 kbps trunks, using packet sizes of 1024 bits, the total delay in transmitting the entire file is

$$\frac{1024}{56000} (4 + 1) + \frac{198976}{56000} = 35.79 \text{ sec.}$$

Cell-switching delay best resembles packet-switching delay. Data exceeding the available throughput is discarded, with no retransmission.

### Impact of Delay on Application

A **bandwidth-limited application** occurs when the receiver begins receiving data before the transmitter has completed transmission of the burst. The lack of bandwidth to hold the transmission limits the transmitter from releasing the entire message immediately.

A **latency-limited application** occurs when the transmitter finishes sending the burst of data before the receiver begins receiving any data. The latency of the response from the receiver limits additional transmission of information.

By applying the basic M/M/1 queueing theory, the average M/M/1 queueing-plus-transmission delay in the network is

$$\frac{b}{R} (1 - \rho)$$

where b: burst length; R: peak transmission rate of the network; \(\rho\): average trunk utilization.

The point where the average queueing-plus-transmission delay exactly equals the propagation delay is called the **latency/bandwidth crossover point**.
Impact of Loss on Application

- For many applications, the loss of a single frame or cell results in the loss of an entire packet because the higher-layer network protocol will fail in attempts at reassembly.
- Loss (or even excessive delay) can result in a time-out or negative acknowledgement in a higher-layer protocol (TCP).
- If the round-trip time is longer than the application window size, then the achievable throughput is greatly reduced.
- The amount of buffering required in the network is proportional to the product of the delay and bandwidth.
- **Go-Back-N retransmission**: all information (N packets) that was sent after the detected error or time-out is retransmitted. This protocol is simple but greatly reduces the throughput.
- **Selective-Reject retransmission**: only the information that was actually in error or timed out is selectively retransmitted. This protocol is complex but more efficient.

The number of cells in the retransmission window is given by

\[ W = \left\lceil \frac{2tR}{\pi p} \right\rceil \]

- \( t \): propagation delay, \( R \): transmission rate; \( p \): packet size in bytes; \( \lceil x \rceil \): ceiling function.
- The probability that an individual packet is lost due to a random cell loss is approximated by \( \pi = \frac{p}{4t} \cdot CLR \).
- The usable throughput for “Go-Back-N retransmission” is approximated by \( \eta_{GBN} = \frac{1 - \pi}{1 + \pi} \cdot W \).
- The usable throughput for “Selective-Reject retransmission” is approximated by \( \eta_{SR} = 1 - \pi \). This formula is valid for the case in which only one packet needs to be transmitted with the round-trip delay window.

Usable Throughput versus Cell Loss Ratio

Data Services Delay

- Greater delay is incurred when an entire message must be read into a node before being transmitted (e.g. frame relay). The user should be informed how many packets or frames (PDUs) can be buffered and at what size (including CRC check).
- Some combination frame/cell switches eliminate this delay by segmenting the frames into cells and immediately transmitting them across the WAN. But, segmentation causes additional overhead in proportional to the frame size.
- Most services, such as frame relay, will state a guaranteed delay objective. For example, 90 percent of PDUs delivered within 1 ms, the rest delivered within 5 ms.
- The effects of protocol conversions, encapsulations, and translations on delay must all be accounted for by the network designer.
7. Availability and Reliability

- Availability and reliability are two quality measures of hardware and software. Their values are usually found through vendor-provided calculations such as Mean Time Between Failures (MTBF) and Mean Time To Repair (MTTR).
- MTBF is calculated based on stress tests, the results of which are projected into the future, as well as through theoretical model projects, possibly using compilations based on the individual parts that make up the system.
- MTTR represents the time it takes to repair the problem or outage when a problem occurs. It is usually stated in minutes.
- Mean Time To Respond (MTTR) is sometimes calculated by service providers, and should not be confused with repair.
- Mean Time To Isolate (MTTI) represents the time it takes to identify the cause of the problem or outage.
- Mean Time Between Service Outage (MTBSO) represents the time since the system has been down for a service outage.

Availability

- Availability is the amount of time the system is working when compared to the measured lifetime of the system, i.e.
  \[ A = \frac{\text{Time the system is working}}{\text{Time system exists between failures}} = \frac{\text{MTBF}}{\text{MTBF} + \text{MTTR}} \]
- For highly reliable systems, the value of \( A \) should be at least 0.999, or 99.9 percent.
- Each additional nine increases the order of magnitude by 10, thus an increase from 99.99 percent to 99.999 percent is a drastic increase in availability.

Unavailability

- Unavailability is a calculation of the time the system will be unavailable or, in other words, its probability of failure.
  \[ U = 1 - A = \frac{\text{MTTR}}{\text{MTBF} + \text{MTTR}} \]  
The system is unavailable \( \text{MTTR} \) hours out of every \( \text{MTBF} + \text{MTTR} \) hours.

Another way to look at the number of failures during a given operation period (say \( t \) hours) is by the formula:

\[ \text{Average number of failures in } t = \frac{1}{\text{MTBF} + \text{MTTR}} \cdot \frac{1}{\text{MTBF}} \]

Example: If \( \text{MTBF} = 1000 \) hours and \( t = 1 \) year, there would be the likelihood of 8.76 failures that year, or 0.024 failures/day.

- For a serial network, the availability and unavailability with two devices would be calculated as:
  \[ A_s = A_1 \cdot A_2 \quad \text{and} \quad U_s = 1 - A_s = U_1 \cdot A_1 \cdot U_2 \cdot A_2 + U_1 \cdot U_2 \]
  - Thus, the greater the number of nodes, the greater the chance of network failure.

- For a parallel network, the availability and unavailability with two devices would be calculated as:
  \[ U_p = U_1 \cdot U_2 \quad \text{and} \quad A_p = 1 - U_p = A_1 + A_2 - A_1 \cdot A_2 \]
  - Thus, the greater the number of nodes, the less the chance of network failure.
Reliability

Reliability is the distribution of time between failures. A high-reliability figure means that the system contains many reliable components which together constitute a reliable system.

Reliability is specified as the probability that the system does not fail (Markovian) prior to t hours:

\[ R = \exp \left( -\frac{t}{MTBF} \right) \]

For a serial hybrid network, the reliability with three devices could be calculated as:

\[ R = \exp \left( -\frac{t}{MTBF_1} \right) \cdot \exp \left( -\frac{t}{MTBF_2} \right) \cdot \exp \left( -\frac{t}{MTBF_3} \right) \]

Example: if we had MTBF figures of 20000, 25000 and 30000 hours, respectively, measured over one year, the network would yield a 34 percent reliability.

Thus, reliability can also be cumulative, but is always as weak as the weakest network element and always decreases as network devices are added.

Plan for Failures

- Make sure that the system is designed to survive failures.
- Select hardware and software with high availability and reliability figures.
- Implement designs that minimize weak failure points by adding additional redundant subsystems and implementing network meshing.
- Always periodically test your backup hardware and systems.

Additional Performance Measurements

- Less than x packets, frames, PDUs among y are delivered without error
- Delivered to wrong customer
- Lost or duplicated
- Total lost frames/packets/cells per time period
- Total network delay
- Guaranteed aggregate throughput per channel based on characteristics of data
- Guaranteed overhead limits (based on packet/frame/cell size)
- Measure of digital data service (DDoS) in error-free seconds
- Load-balancing/load-sharing limitations
- Masking, filtering and forwarding rates for access & backbone
- Error detection/correction effects on overhead & throughput
- Level of redundancy built into network

Significant Failure, e.g. failure of an entire switch

- The general guideline is that any element of the network should not become isolated by a single trunk or switch failure.
- This can be solved with equipment diversity, circuit diversity, or dial or dedicated backup facilities.
- Different performance objectives under failure situations.
- Some traffic can be preempted during a failure scenario so that support for mission-critical traffic is maintained.

Traffic Overload & Unexpected Traffic Patterns

- Traffic overload and unexpected traffic patterns can cause congestion, which can greatly reduce the overall throughput.
- Dynamic Flow Control: detect congestion, isolate sources, provide feedback to throttle the sources in a fair manner
- Switched Virtual Channel: reserve bandwidth for exclusive use
9. Network Performance Modeling

- **Assumption**: nodes operate independently, and the traffic mixes and splits independently and randomly.
- **Simulation**: very useful in investigating the detailed operation of a system, often take too long to execute.
- **Analysis**: less computationally intensive, inaccurate.

The inputs of a network model are topology (graph, characteristics of nodes and links), traffic (pattern between nodes), and routing (set of links → path).

The outputs are measures of performance (loss and delay statistics) and cost. The network designer need to select an effective price-performance tradeoff.

10. Creating the Traffic Matrix

- **Traffic matrix** contains the information that who needs to talk with whom, and what traffic patterns need to go where. It is, in effect, a spreadsheet or table mapping traffic flows from origin to destination.

**Asymmetric versus Symmetric Distribution**

- **Asymmetrical traffic** lacks directional symmetry through an imbalance of flows, speeds, or other characteristics. It originates from large sites to small, or vice versa, and does not follow a normal distribution. Access devices vary in quantity, design, engineering, and loading.

- **Symmetrical traffic** often originates from communities of similar interest, such as specific geographic regions, and is uniformly spread across these sites within each region and the bidirectional quantity is similar. Most of access devices are similar in quantity, design, engineering, and loading.

**Creating the Traffic Matrix**

- Access node/point: could be a concentrator (e.g. PAD or FRAD), Digital Cross-Connect (DXC), access multiplexer, hub, bridge, router, access switch, or any other device concentrating user inputs but not operating as a switching backbone.

- The traffic matrix helps define details about the access node requirements, such as location, size, operation, protocol support, performance characteristics, and device type.

- Start with a local geographic area as node A and work out from there, so that letters that are lexically close represent nodes in a geographic region.

- Traffic measurements used should have a similar message (packet) length and should be measured from either the network or the application point of view, and in “mean busy hour” throughput (in bps) or in the peak-to-mean ratio.

- Mark invalid and preferred routes.
**Interpreting the Matrix**

- In its simplest use, the matrix shows the connectivity required between each site.
- All traffic identified as remaining local to a given node would be placed into the same access node.
- Group nodal traffic distributions in a local geographical area together to form larger access nodes.
- The small amount of traffic originating at some nodes is “backhauled” to the regional concentrator.
- This process continues until the number of access nodes required to begin the design is established.
- Network designs for multimedia and multiprotocol networks are much more complicated. These designs often require many traffic matrices combined into a multidimensional matrix (e.g., a z-axis to represent priority or protocol), or in large networks, design tools to perform these calculations.

**Capacity Planning and Network Vision**

- A short-term objective and task-oriented plan is usually revised each year, and the long-term 5- to 5-year plan should also take into account the strategic vision of corporate communications for the next 5 to 10 years. Both plans must take into account the business needs, customer needs, and the technologies.
- As the cost of computing hardware decreases, the entropy of capacity requirements increases. This makes capacity planning for communications networks a challenging task.
- Short- and long-range capacity planning can provide the corporation with a competitive advantage, assuming that the plans fit in with the long-range strategic business plan.
- Each design must provide the flexibility to change technology as business needs change without major network restructuring.
- One of the best architectures is that which is built in a hierarchical nature that caters to flexible peer-to-peer communications and can be replaced in layers over time.