# VolP

Voice over Internet Protocol



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- Using VoIP we can easily call anywhere in the world where internet connection is available.
- VolP systems usually interface with the traditional public switched telephone network (PSTN) to allow transparent phone communications worldwide.
- This system has potential to change completely current phone system.

#### What is VoIP?

 VoIP (Voice over Internet Protocol) is simply the transmission of voice traffic over IP-based networks (internet/intranet).

VoIP is a method for taking an analog signal and turning them into digital data so that they can be transmitted over the internet.

#### What is VoIP?

- Using VoIP we can turn a standard internet connection into a way to place free phone calls.
- It is also known as
  - > IP telephony
  - Internet telephony
  - Voice over Broadband (VoBB)
  - Broadband phone

# How traditional phone system works?

- ☐ Before knowing How VoIP works, first let's understand how current system works.
- Current system works on circuit switching technology.
- When u dial a number you are connected to that number using existing PSTN (Public Switched Telephone Network).
- The dedicated connection is being made between two phones, which is maintained for the duration of the call.

 This is the simplest diagram which represents VoIP transmission.



- It's a three step process
  - Source side Processing
  - 2. Transmission over Network
  - 3. Reconstruction at Receiver

- Step 1: Source side Processing
- ADC : An Analog voice signal is being converted into Digital signal using ADC.
- Compression Technique: An compression algorithm is being applied on digitized signal. Data compression is a process whereby voice data is compressed to render it less bulky for transfer over network.
- The compression used in VoIP is lossy compression, in which some of the elements of the audio stream is lost - we may loss some information.

- Step 1: Source side Processing
  - But this loss does not much affect the quality because it discards the sound that cannot be heard by human ear which is useless to be transmitted.
  - Also silence is discarded.
  - > For this compression we have different CODECs (coder decoder).
  - Codec is simply an algorithm which is installed on server which have ATA or IP phone connected to it.
  - Compression software (called a codec) encodes the voice signals into digital data that it compresses into lighter packets that are then transported over the Internet

#### Step 1: Source side Processing

Common VoIP Codecs (Reference only)

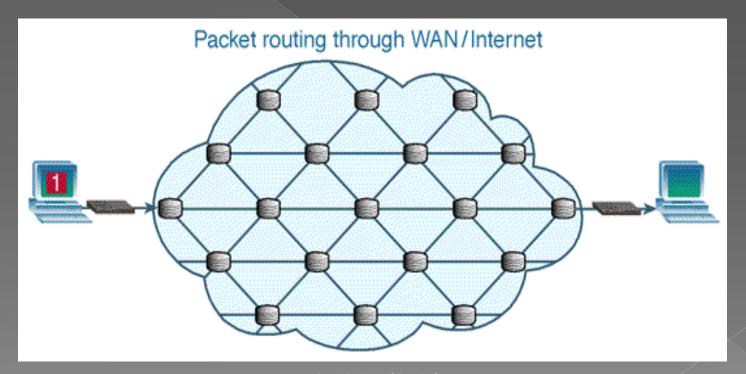
| Codec   | Bandwidth/kbps | Comments  |
|---------|----------------|---|
| G.711   | 64             | Delivers precise speech transmission. Very low processor  |
|         |                | requirements. Needs at least 128 kbps for two-way.  |
|         |                |   |
| G.722   | 48/56/64       | Adapts to varying compressions and bandwidth is conserved with network congestion.  |
|         |                |   |
| G.723.1 | 5.3/6.3        | High compression with high quality audio. Can use with dial-up. Lot of processor power.   |
| G.726   | 16/24/32/40    | An improved version of G.721 and G.723 (different from G.723.1)   |
| G.729   | 8              | Excellent bandwidth utilization. Error tolerant. License required.  |
| GSM     | 13             | High compression ratio. Free and available in many hardware and software platforms. Same encoding is used in GSM cell phones (improved versions are often used nowadays). |
| iLBC    | 15             | Robust to packet loss. Free   |
| Speex   | 2.15 / 44      | Minimizes bandwidth usage by using variable bit rate.   |
|         |                | Courtesy: http://yoip.about.com/od/yoiphasics/a/yoir  |

- Step 2 : Transmission over Network
  - Packet switching technology is being used for routing packets over internet.
  - Packet is one type of data record which contains the receiver's and sender's IPaddress, data & packet number.

| Header  | Sender's IP address<br>Receiver's IP address<br>Protocol<br>Packet number | 96 bits  |
|---------|---|----------|
| Payload | Data  | 896 bits |
| Trailer | Data to show end<br>of packet<br>Error correction                         | 32 bits  |

Packet (1024 bits)

Step 2 : Transmission over Network



Packet Switching

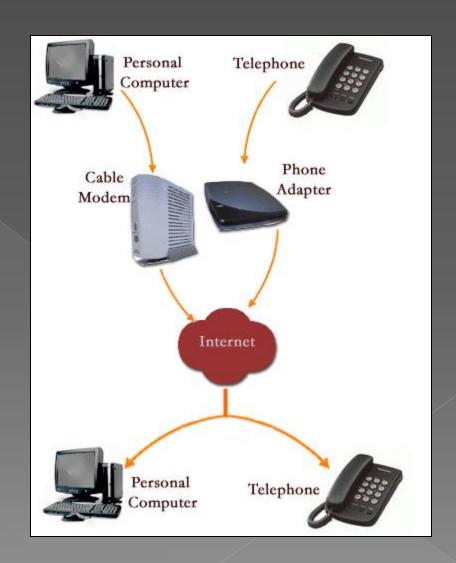
- Step 2 : Transmission over Network
  - Packet Switching

| Advantages              | Disadvantages                   |
|-------------------------|---------------------------------|
| Efficient use of n/w    | Latency-delay in packet routing |
| Multiple available path | Packet loss                     |
| Network scalability     | Order less packet routing       |

- Step 3 : Reconstruction at Receiver
  - > The packets received may be in random order.
  - > To reorder them, one buffer is used in which the packets are stored and then they are sequentially ordered.
  - Then this packets are send to an D/A converter & lastly we get the analog o/p (voice) at receiving end.

# Using VoIP

- For connecting to VoIP network, we have different methods.
- 1. Using your computer.
- 2. Using simple telephone with phone adapter.
- 3. Using VoIP telephone



# Advantages

- Low cost
  - Because in IP-telephony only packets are send and received, while in traditional system the connection is made until end of call.
- Increased functionality
  - You can connect your phone from anywhere to network and receive your incoming calls.

# Disadvantages

- VolP is dependent on wall power.
  - > If your power goes out the VoIP phone will not work. While current phone system works on phantom power.
- 911 Emergency call
  - VoIP phone uses an IP address as a phone number & we can find the geographical position using that IP – address.
  - To fix this, perhaps geographical information could somehow be integrated into the packets.

# Disadvantages

#### Latency

- Latency is the time between the moment a voice packet is transmitted and the moment it reaches its destination.
- > This leads to delay & echo which is undesirable in voice communication.

#### Jitter

Unfortunately, the delay is not always constant, and varies depending on network availability. This variation in delay is called jitter, which causes damage to voice quality.

# Quality of Service(QoS)

- QoS depends on following factors
  - > Type of internet connection
  - VoIP hardware
  - Codec: compression techniques used
  - Location of hardware
    - If your ATA is too close to your broadband router, you might experience voice quality problems.

### References

- www.about.com
- www.howstuffworks.com
- www.wikipedia.com

# Thank You

Questions ???