Lecture – 14 (Dt. 15th April 2020)

Electronic Switching (EC- 8th Sem)

Voice Over Internet Protocol (VOIP)

References :

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Voice over Internet Protocol, also called Voice over IP or just VoIP technology is having a major impact on the telecommunications industry. VoIP technology provides advantages for both the user and also the provider, allowing calls to be made more cheaply, as well as enabling data and voice to be carried over the same network efficiently. In view of the way VoIP technology is being adopted, telecommunications providers are having to adopt the new technology. Already it has caused some impact on major businesses, and there will be more to come.

Until recently voice traffic was carried using a circuit switched approach. Here a dedicated circuit was switched to provide a call for a user. Now with new data and Internet style technology used for VoIP, packet data and Internet Protocol (IP) is used to enable a much more efficient use of the available capacity.

What is VoIP?

- *VoIP definition:* VoIP (voice over IP) is the transmission of voice and multimedia content over Internet Protocol (IP) networks. VoIP is enabled by a group of technologies and methodologies used to deliver voice communications over the internet, enterprise local area networks or wide area networks.

The concept of Voice over Interent Protocol, Voice over IP, or VoIP, is quite straightforward. A VoIP system basically consists of a number of endpoints which may be VoIP phones, mobile phones, VoIP enabled browsers on computers, etc and also an IP network over which the packet data is carried.

In a VoIP system, the phone or computer acting as an endpoint consists of a few blocks. It includes a vocoder (voice encoder / decoder) which converts the audio to and from the analogue format into a digital format. It also compresses the encoded audio, and in the reverse direction it decompresses the reconstituted audio. The data generated is split into packets in the required format by the network interface card which sends them with the relevant protocol into the outside world. Signalling and call control is also applied through this card so that calls may be set up, pulled down, and other actions may be undertaken.

The IP network accepts the packets and provides the medium over which they can be forwarded, routing them to their final destination. As complete circuits are not dedicated to a given user, at times when no data needs to be sent, for example during quiet periods in speech, etc., the capacity can be used by other users. This makes a significant difference to the efficiency of a system, and allows significant savings to be made.

Traditionally the term VoIP referred to systems where IP was used to connect private branch exchanges, PBXs, but now the term is more widely used and encompasses IP telephony.

VoIP Protocols

In order to be able to communicate using a VoIP system, there are a number of protocols that may be used.

- *H323:* The signalling protocol is used to control and manage the call. It includes elements such as call set up, clear down, call forwarding and the like. The first protocol to be widely used for VoIP was H323. However this is not a particularly rigorous definition and as a result other variants have been developed.
- *Skinny:* One other signalling protocol that was used was known as "Skinny" and is a Cisco Proprietary protocol and is from Nortel and another is called Unistem. In view of this there are often interfacing problems.
- *SIP:* SIP, Session Initiation Protocol, is now being widely adopted as the main standard is a far more rigorous protocol for signalling and is the one that is most widely used now.

- *RTP:* RTP, Real Time Protocol, is a data exchange protocol and this can handle both audio and video. RTP handles the data exchange, but in addition to this a codec is required. Where voice is used a vocoder is used (a codec can be used for any form of data including audio, video, etc).
- *G711:* G711 is possibly the most widely used VoIP vocoder and it is the standard for transmitting uncompressed packets. G.729 is the standard for compressed packets. Many equipment vendors also use their own proprietary codecs. Voice quality may suffer when compression is used, but compression reduces bandwidth requirements. There are many other vocoders/ codecs that are used with varying data rates and providing different levels of voice quality.

Service quality

Quality of Service, QoS, for the data link has a major impact on VoIP perceived sound quality. The data exchange must take place in real time and any delays in the system cause significant disruption to the traffic. Delayed packets may mean that packets arrive out of order, or with varying gaps between them, resulting in garbled speech, Packets may even disappear resulting in lost information.

For any packet passing through an IP network it is possible to define the class of service required. It is important that packets that need to be transferred in real time are given a higher quality of service than those that can be transferred as the network permits. This is particularly important for services like VoIP that are termed delay sensitive applications.

Advantages

Voice over IP, VoIP technology provides a number of significant advantages to operators and to users. For the user one of the main advantages is the flexibility. Phones are software based, sometimes being attached to computers. As a result a considerable degree of flexibility is afforded to the user. It is possible to move the phone around and by enabling the system to recognise the individual phone it is possible to route the data to it automatically. In addition to this ideas such as mobile IP could enable the user to be located away from the home network and still receive calls.

A further advantage is that the wireless network technologies such as 802.11 can carry the calls as voice is simply another form of application. This gives further flexibility as the phone does not have to be physically wired to a network. Again Quality of Service is a major factor and this is being addressed under 802.11e

For the operator some of the advantages are different. One of the major drivers towards the use of VoIP is cost. Previously digital traffic was handled using time division techniques. This had the disadvantage that when a particular time slot allocated to a user was dormant, it could not be used. Using IP techniques much higher levels of efficiency can be attained. Although the system required to carry packet data is more complicated, the returns far outweigh the additional costs.

As with all technologies there are disadvantages. The main one with VoIP is voice quality. This results from the use of a vocoder to digitise and compress the audio. Quality is comparable with that from a mobile phone, but for the future with rapidly improving standards of vocoders there are likely to be significant improvements in this area.

In the long term VoIP is the way the market is moving, and now with increasing speed. Offering not only great improvements in flexibility, but also major cost savings, but with the requirement for large levels of investment, this is the way that the telecommunications market is moving. However to remain competitive it will be necessary to adopt the new VoIP technology.

VoIP protocols overview

Although working together, there are a number of different organizations and bodies that are mentioned when referring to VoIP protocols:

- IETF This is the Internet Engineering Task Force. It is a community of engineers that defines some of the prominent standards used on the Internet (including VoIP protocols) and seeks to spread understanding of how they work.
- ITU the International Telecommunication Union. This is an international organization within the United Nations System used by where governments and private sector companies to coordinate and standardize telecommunications networks, services and standards on a global basis.

In addition to the organizations involved, there is also a variety of different VoIP protocols and standards.

- H.248 H.248 is an ITU Recommendation that defines "Gateway Control Protocol" and it is also referred to as IETF RFC 2885 (Megaco). It defines a centralized architecture for creating multimedia applications and it extends MGCP. H.248 is the result of a joint collaboration between the ITU and the IETF and it is another VoIP protocol.
- H.323 This is ITU Recommendation that defines "packet-based multimedia communications systems." H.323 defines a distributed architecture for multimedia applications, and it is thus a VoIP protocol.
- Megaco This is also known as IETF RFC 2885 and ITU Recommendation H.248. H.248 defines a centralized architecture for creating multimedia applications.
- Media Gateway Control Protocol (MGCP) This is also known as IETF RFC 2705. It defines a centralized architecture for creating multimedia applications, and it is therefore a VoIP protocol.
- Real-Time Transport Protocol (RTP) This VoIP protocol is defined under IETF RFC 1889 and it details a transport protocol for real-time applications. RTP provides the transport mechanism to carry the audio/media portion of VoIP communication and is used for all VoIP communications.
- Session Initiation Protocol (SIP) This is also known as IETF RFC 2543 and it defines a distributed architecture for creating multimedia applications.

Centralised and distributed architectures

One of the advantages of VoIP is that it does not legislate for the architecture of the network that carries the data. Early telecommunications networks used a centralised structure where all the intelligence was contained at the switching station or exchange. With the advent of packet technology, the routing and intelligence can be distributed to where it is most convenient to locate it. This may be by having a distributed architecture, or a centralised one. While both architectures can be employed with VoIP, the type of architecture does have an impact on the optimum VoIP protocols to use. This is one of the reasons why a number of VoIP protocols are used, and will remain to be used.

P networks carrying VoIP traffic are very complicated. They carry both voice and data traffic and this results in a variety of traffic with different requirements being carried and this presents many challenges. In order to ensure that all the requirements are met and the network operates to its maximum efficiency can present many challenges. Obviously the design must be correct, but once implemented testing of the network is needed to ensure that it is able to operate correctly when installed, and then maintained correctly ensuring that its performance is maintained or optimised to provide the performance meets the needs of the network provider and the user. For VoIP, testing is an essential element of any network. However specialised VoIP testing techniques are required.

VoIP network architecture

The structure of a VoIP network comprises many entities and this means that VoIP testing is essential to ensure that the network is operating satisfactorily. A typical VoIP network will include many different entities:

- Signalling gateways
- Media gateways
- Gatekeepers
- Class 5 switches
- SS7 network
- Network management system
- Billing system

This variety of different entities within the VoIP network all communicate with each other using a variety of protocols. To perform correctly it is necessary to ensure that they communicate efficiently and that no bottlenecks are created. Analysing the performance of a VoIP network is not always easy. However it can be achieved and significant improvements in performance can be achieved if the VoIP testing scenarios are carefully chosen and planned, and the data analysed to reveal any problems.