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Emerging New Trends in Transmission of Real Time Application(VOIP) to NGN By RTP

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Abstract: A major consideration with the NGN is that it provides an equivalent voice service quality and resiliency as the current telephone network. The analysis carried out for this paper looked that without making any real time arrangement in data network in case of IP network will definitely detorate the Quality of service. So arrangement of Real time is mandatory for network quality and QOS of voice. In case of VOIP it is simply a transmission of voice packets over layer 3 i.e. Network Layer .No fix quality measures has been taken so far to achieve the quality in VOIP. By using the RTP& RTCP as a Application Layer Signaling Control Protocol in NGN network will ensure the bare minimum packet loss(pay load) for telephony. The scope of this paper will prove that real time application i.e. voice transmission if it sent over non real time network with RTP/RTCP(Application Layer Signaling Control Protocol) the quality increases & provides an equivalent voice service quality and resiliency as the current telephone network.

Keywords:- NGN, VOIP, RTP, RTCP, QOS, IP.

I. INTRODUCTION

The NGN concept takes into consideration new realities in the telecommunication industry characterised by factors such as: the need to converge and optimise the operating networks and the extraordinary expansion of digital traffic i.e., increasing demand for new multimedia services, increasing demand for mobility, etc. The customers demand for new services is increasing and that too at less cost. Therefore there is a need for a network which has a capability to develop services and able to extend it to the end user independent of the other part of the network. This is achieved through the concept of NGN[11].

NGN Architecture

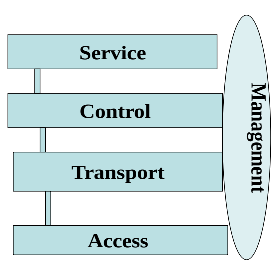

Fig 1

Fig 1 depicts the basic architecture of NGN.as we can see it is a frame work only.All the four layers of network is controlled by single management layer.Further To change over from current generation network to next generation network we have to move in a step-by-step manner to safeguard our existing network infrastructure and investment and therefore we have to follow an evolutionary path.

In NGN basically the call control (i.e. signaling) and the switching is separated out in different layers and between these layers open interfaces are used. The call control functionality is realised by the component which is called call server or softswitch or media gateway controller and the interfaces to the existing PSTN switches is done with the help of media gateways for voice transport and by signaling gateways for signaling transport. For switching and transport of the packets existing IP/MPLS backbone is used. With NGN architecture the new and innovative services can be given very fast and cost effectively. Also the capital expenditure and operational expenditure come down drastically[13].

NGN concept with respect to voice
For voice model the basic changes in present circuit based central office switches are call control will be replaced by call server,switching will be done by MPLS network and present interfaces will be done by various gateways,as shown in fig.2.

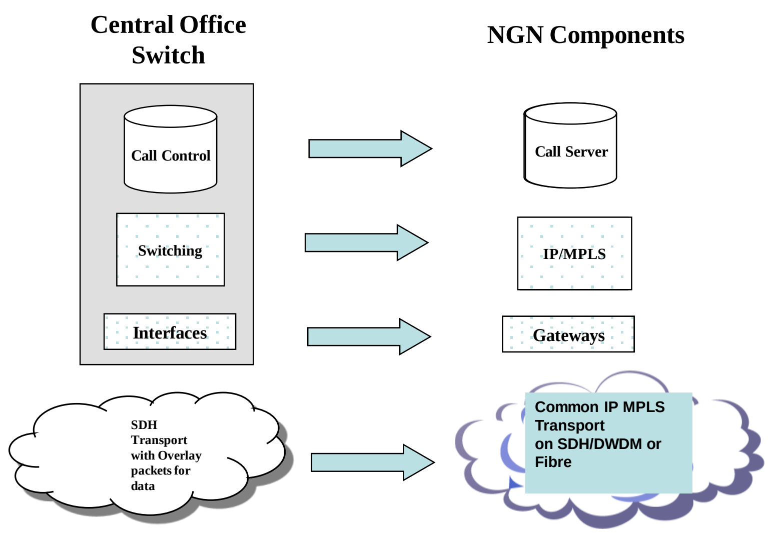

Fig2

The NGN is characterised by the following fundamental aspects:

- Packet-based transfer
- Separation of control functions among bearer capabilities, call/session, and application/service
- Decoupling of service provision from transport, and provision of open interfaces
- Support for a wide range of services, applications and mechanisms based on service building blocks (including real time/streaming/non-real time services and multi-media)
- Broadband capabilities with end-to-end QoS and transparency
- Interworking with legacy networks via open interfaces
- Generalised mobility
- Unfettered access by users to different service providers
- A variety of identification schemes which can be resolved to IP addresses for the purposes of routing in IP networks
- Unified service characteristics for the same service as perceived by the user
- Converged services between Fixed and Mobile networks
- Independence of service-related functions from underlying transport technologies
- Support of multiple last mile technologies
- Compliant with all Regulatory requirements, for example concerningemergency communications & security/privacy, etc.

Evolution of Present Networks to NGN:Related Issues
The high capital investment in the PSTN (public switched telephone network) means it must be one of the main work areas for studying evolution to the NGN. The resulting outputs describe possible ways of evolving the PSTN to become an NGN.

1) QoS (Quality of Service)

The basic criterion for QoS evolution is ‘subjective user satisfaction’, e.g. speed, accuracy, reliability, and security. This involves identification of parameters that can be directly observed and measured at the point at which the service is accessed by users and network providers. Flexibility within the global end-to-end NGN architecture is essential to allow for each recognised operating agency’s different regulatory environment, service offerings, geographic span, and network infrastructure. These factors need to be taken into account when agreeing on parameters for, and levels of, QoS for NGN.

2) **Interoperability**
Considering that the NGN will involve a broad series of protocols (including various profiles) at both service and network levels, it is essential to ensure interoperability between different systems and networks.

3) **Security**
Security is as crucial to the NGN as it is in today's network environment. The very wide scope of this topic, combined with the number of SDOs (standardsdevelopment organisations) already involved, underlines the strategic importance of this subject. Within the NGN, security issues interrelate with architecture, QoS, network management, mobility, charging and payment.

4) **Generalized Mobility**
NGN will give users and devices the ability to communicate and to access services irrespective of change of location or technical environment. The degree of service availability may depend on several factors, including access network capabilities, service level agreements between the user's home network and visited networks, etc. It includes the ability to communicate from various locations using a variety of terminal equipment, with or without service continuity while in transit or while changing access means. This includes recognition of the need to converge the previously distinct worlds of fixed and mobile telecommunications into a coherent whole.

5) Service Capabilities and Architecture
Needs to develop a suitable service architecture focused on the interfaces to support different business models and seamless communication in different environments to provide the service capabilities of existing one[12].

VOIP

It is a possibility of voice transmission over a IP layer (Network Layer. In 1999–2000, VoIP was one of the most successful buzzwords of the telecom . Every start-up company had a 'new service' or a 'killer application' that would change the landscape of the telecommunication industry for ever

VoIP Quality of Service

Voice traffic differs from data traffic in the following ways:
• Data is often bursty by nature; voice is deterministic (smooth).
• Data applications resend dropped packets; voice applications can only conceal dropped packets.
• Data applications can usually tolerate some delay; voice applications must minimize delay, so that the recipient does not hear clips in the transmission[5].

Delay

Delay is the time it takes for VoIP packets to travel between two endpoints. Because of the speed of network links and the processing power of intermediate devices, some delay is expected; however, you should attempt to minimize this delay.The human ear normally accepts a delay of about 150 milliseconds (ms) without noticing problems. (The ITU G.114 standard recommends no more than 150 ms of one-way delay.) When delay exceeds 150 ms,a conversation becomes more and more like a citizens band (CB) radio interchange in which one person must wait for the other to stop speaking before beginning to talk. This type of delay is often evident on international long-distance calls. You can measure delay fairly easily by using ping tests at various times
of the day with different network traffic loads. If network delay is excessive, reduce it before deploying VoIP in your network[14].

Jitter

Although delay can cause unnatural starting and stopping of conversations, variable-length delays (also known as *jitter*) can cause a conversation to break and become unintelligible. Jitter is not usually a problem with PSTN calls because the bandwidth of calls is fixed. However, in VoIP networks in which existing data traffic might be bursty, jitter can become a problem.

Serialization

Serialization is a term that describes what happens when a router attempts to send both voice and data packets through an interface. In general, voice packets are very small (80 to 256 bytes), and data packets can be very large (1500 to 18,000 bytes).

Problems associated with VOIP QOS

. It is always of a matter of concern that quality is poor over IP layer especially in case of Voice. The main reason behind that is we all are using to send the real time application over a non real time network.because IP layer is meant for Data only and we are using it for voice transmission ,we are seeking the Telecom substitute over the IP Network.

II. RTP

RTP is real time transport protocol and it is developed basically for media transport in most of the NGN applications although other Transport Layer protocols, in particular Datagram Congestion Control Protocol (DCCP) and Stream Control Transmission Protocol (SCTP) may be used because of their congestion control mechanisms.

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0                   1                   2                   3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
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|                           timestamp                           |
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|           synchronization source (SSRC) identifier            |
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|            contributing source (CSRC) identifiers             |
|                             ....                              |
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Fig.3

Transmission Control Protocol (TCP) has been documented for RTP usage, but is rarely deployed in such applications [1]. Applications using RTP are less sensitive to packet loss, but typically very sensitive to delays resulting from network latency, so UDP is a better choice than TCP for such applications.

- Payload-type identification - Indication of what kind of content is being carried
- Sequence numbering - PDU sequence number
- Time stamping - allow synchronization and jitter calculations
- Delivery monitoring[6]

III. RELATED WORK

So far thorough work has been done in the analysis of delay and jitter of voice traffic over internet in which thorough assessment of voice delay carried out. It is shown that Priority Queuing is the most appropriate scheduling scheme for the handling of voice traffic, while preemption of non-voice packets is strongly recommended for sub-10 Mbit/s links. also find that per-connection custom packetization is in most cases futile, i.e. one packet size allows a good compromise between an adequate end-to-end delay and an efficient bandwidth utilization for voice traffic. It is also focused on the effect of the residual transmission time of non-voice packets of voice delay,and showed that it constitutes the largest portion of voice delay in case PQ is used to schedule voice traffic.It is also shown that effect of packet size on voice delay and bandwidth utilization,and showed that a packet formation time of 30 ms for G.729A and G.723.1 (that is, a packet size of 30 and 20 bytes, respectively)and 10 ms for G.711 (that is, a packet size of 80 bytes)constitute a good compromise between low delay and efficient network utilization.

It is analyzed that The covert channel modifies the timestamp value in the RTP header to send its secret messages. The high frequency of RTP packets allows for a high bitrate covert channel, theoretically up to 350 bps. The broad use of RTP for multimedia applications, including VoIP, provides plentiful opportunities to use this channel. By using the RTP header, many of the challenges present for covert channels using the RTP payload are avoided.

IV. PROBLEM IDENTIFICATION

The major drawback of this existing network architecture is features. Services are duplicated at each and every access point. Because we are not having Centralized control and distributed architecture. Definitely new services will be expected in the NGN / EOIP such as broadcast media transmission and video on demand. But one major issues are always from past so many years are associated with Internet Protocol regarding voice Transmission[5].

V. PROPOSED WORK

This paper concentrates on improving network quality specially when we send the real time application such as voice over a non real time network i.e. IP network. To identify this and to sort out the solution RTP & RTCP protocol will be deployed in between the two NGN nodes can be Trunk Media Gateways through the VTCA system and traffic will be generated and sent over this IP based network. Through the analysis software it will be identified with the help of RTP/RTCP how much packet loss / delay / jitter reduced up to much extent and the quality increases & provides an equivalent excellent voice service quality and resiliency as the current telephony network. The analysis will be carried out for this paper looked that without making any real time arrangement in data network in case of IP network will definitely detorate the Quality of service. So arrangement of Real time is mandatory for network quality and QOS of voice .By using the RTP& RTCP as a Application Layer Signaling Control Protocol in EOIP network will ensure the bare minimum packet loss(pay load) for telephony.

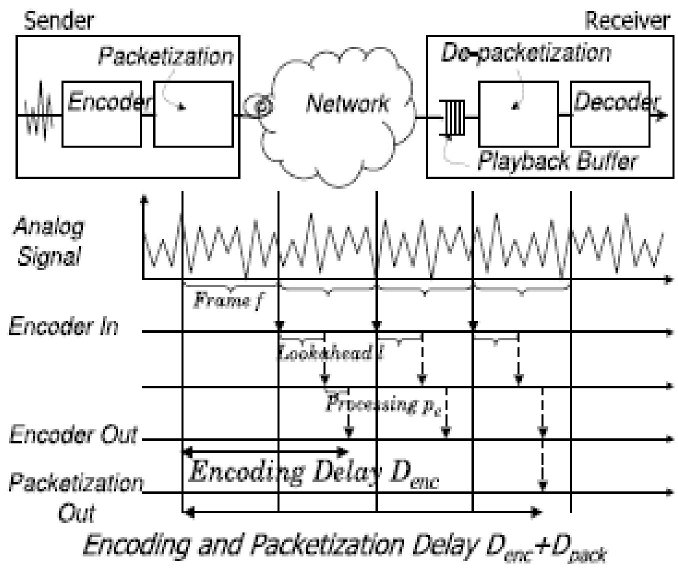

Fig 4

We can focus on networks where a queue for voice traffic is provided. In this context, we have first described the particularity of voice traffic as compared to other traffic on the network i.e. real time need analysis fig 4 . We can show the importance of bandwidth to reduce the delay percentile incurred by voice in a network, and concluded that network delay becomes a negligible portion of end-to-end delay in case available bandwidth exceeds 10 Mbit/s.(Between Media Gateway to Media Gateway) and after that we can compare different scheduling schemes (that is, PQ, Priority Queuing, WRR Weighted Round Robin and the provision of different circuit for voice traffic), and can show that PQ leads to the best compromise between bandwidth utilization and delay minimization, as long as the preemption of non-real time voice packets is implemented on sub-10 Mbit/s links. In last we can analyze the
effect of packet size on voice delay and bandwidth utilization, and showed that a packet formation time of 30 ms for G.729A and G.723.1 (that is, a packet size of 30 and 20 bytes, respectively) and 10 ms for G.711 (that is, a packet size of 80 bytes) constitute a good compromise between low delay and efficient network utilization.

VI. Conclusion

The conclusion of this paper is that RTP defines transport support for common functions of real-time applications. So provision of Real time is quality demand for network and for quality grade service of voice .By using the RTP& RTCP as a Application Layer Signaling Control Protocol in EOIP network will ensure the bare minimum packet loss(pay load) for telephony. Timing information: sampling period. .

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