Abstract

The telephone networks of today are changing from traditionally circuit-based networks to the packet-based networks. This change of network technology opens challenges to new voice technologies. One technology is Voice over IP (VoIP) and three other main technologies are Voice over Asynchronous Transfer Mode (VoA), Voice over Digital Subscriber Line (VoDSL), and Voice over Cable Television (VoCATV).

This paper presents the basic information for these three other technologies. Other point of view on this paper is to examine what is a media path and what is a signal path on these three technologies. A Session Initiation Protocol (SIP) is used on the IP telephony technology and on this paper we go through what signaling protocol is used on the other voice over technologies.

1 Introduction

The VoA is based on cell technology and it is asynchronous with high transfer speed e.g. 622 Mbps. Asynchronous Transfer Mode (ATM) architecture has four different adaptation layers and three of them are used to carry voice. ATM is the first important technology that challenge the circuit based telephone networks. The VoDSL is local loop access technology and transfer speed is up to 9 Mbps. The international markets for DSL technology is one of rapid growing modem technology. Market of VoDSL is growing fast in USA and has been estimated that VoDSL markets of USA will grow about 75 % this year (Adtran Inc.).

Community Antenna Television (CATV) is unidirectional from server to customer site and VoCATV is based on interactive services. The CATV operators have changed CATV network topology from unidirectional to bi-directional to offer these new interactive services to customers. There are many groups or associations who have done different specifications from this area. Main target is to offer to end user better services with one interface from a residence or from a small office. The CATV system can be used to transfer TV programs, to carry telephony signals, and to browse Word Wide Web (WWW) pages.

2 Voice over ATM

2.1 ATM networks

ATM network is based on packet switching technology and it is used on the telecommunications networks and in data communications networks. International Telecommunication Union (ITU) has selected ATM technology for realising a Broadband-Integrated Services Digital Network (B-ISDN) [1]. There are different limitations on traditional telecommunications networks, networks like Plesiochronous Digital Hierarchy (PDH) and Synchronous Digital Hierarchy (SDH). Traditional networks were service dependence, inflexible and inefficient. Different networks were designed to support a particular service. PDH networks’ and SDH networks’ main use was voice carry and CATV network was designed to carry TV channels. Other equipment on traditional networks were designed to use specific bandwidth like 64 kbps and its' multiplies e.g. 32*64 kbps. Today telecommunications networks are based on circuit switching. A phone call reserves 64 kbps point-to-point channel through a telecommunications network and this channel or path is the used only for this phone connection.

For these limitations on networks ITU started standardisation work in the late 1980s, known as B-ISDN. ATM was chosen as the transfer technology for B-ISDN. ATM technology is service-independent and based on 53-byte fixed-size cells. Each cell has 5 bytes header and 48 bytes payload, so ATM network's efficiency is about 90.6 %. ATM transfer mode has five ATM Adaptation Layer (AAL) types. These five layers are used to carry diverse message streams all with same ATM cell format.

- AAL Type 1: Constant Bit Rate (CBR) Services; for isochronous information streams like voice (64 kbps), uncompressed video and leased lines.
- AAL Type 2: Variable Bit Rate (VBR) Services; audio and compressed video (Motion Picture Experts Group 2, MPEG2)
• AAL Type 3/4: Connection-oriented VBR Services; Data Transfer on packet switching networks
• AAL Type 5: Connectionless VBR Services; Data Communications over Transmission Control Protocol/Internet Protocol (TCP/IP).

2.2 Voice over ATM

The ATM Forum [2] and its group, Voice and Telephony Services over ATM (VTOA) [3], have published since 1995 different specifications, for voice transport on ATM networks. Figure 1 describes different protocols that can be transferred via ATM using AAL1 layer and using AAL2 layer. In the same figure there is described control traffic over ATM. ATM forum has published specification named “Voice and Telephony Over ATM to the Desktop” [4], where is described voice packetization to AAL5 cell.

Figure 1: Inter-PSTN over ATM trunking Protocol Stack [5]

2.2.1 User plane

ATM trunking which uses AAL2 for narrowband services specification (AF-VTOA-0113.000) [6] describes transport of narrowband services across an ATM network between two Interworking Functions (IWF) to interconnect pairs of non-ATM trunks. ATM trunking using AAL2 offers some benefit compared with other trunking methods; bandwidth allocation is less per cell, silence removal releases bandwidth and operator can route and switch narrowband calls on a per call basis. AAL2 layer is better for bursty traffic like voice communications because this layer uses VBR transport on ATM networks. Figure 1 describes techniques and the system can map bursty traffic better to VBR traffic than to CBR traffic on AAL1 layer. Silence is a normal occurrence in phone call, transmission is a duplex operation and sometimes both speakers are silent, one speaker is silent or speech has silent intervals. When this silence happens on speech, the system does not transmit encoded speech information or empty ATM cells to the other user. The methods used to detect silent periods are specified in the ITU-T I.366.2 specification. The routing decision can be based on information of IWF, characteristics of the call as expressed in the received Signaling or incoming interface and timeslot. Other main service to IWF module is mapping Time Division Multiplexing
(TDM) frame or Tributary Unit (TU) signals to ATM cells and vice versa.

What are the difference between these three layers; AAL1, AAL2, and AAL5? AAL1 packet has 47 octets reserved to payload of voice and only one octet is reserved to voice header. AAL1 voice header contains Convergence Sublayer Indicator (CSI) bit, Sequence Count (SC), Cyclic Redundancy Check (CRC) control, and parity bit. Ordinarily one 64 kbps voice channel, e.g. ISDN or Pulse Code Modulation (PCM), is mapped to one ATM Virtual Channel Connection (VCC) and it is named one-to-one mapping, but it is possible that one VCC curries multiple 64 kbps channels and it is named many-to-one mapping. AAL2 packet has 48 octets payload and this packet can include several frames of users. AAL2 has two parts that have their own tasks, a Service Specific Convergence Sublayer (SSCS) and a Common Part Sublayer (CPS). The SSCS has two different elements for data and for voice. Segmentation and reassenby element is for data (I.366.1) and trunking element is for voice (I.366.2). The CPS packet has 3 octets packet header and CPS info part that can be from 1 octet frame to 45/64 octets frame. AAL5 packet has 40 octets voice payload part and 8 octets trailer. The trailer part includes Common Part Convergence Sublayer User-to-User indicator (CPCS-UU), Common Part Indicator (CPI), and CRC with 32 bytes. AAL5 packets are used with VTOA desktop terminals.

Delays and echo are important variables on speech quality. ITU-T recommendations G.114, G.131, and G.126 specify the average national network delay for speech traffic. Acceptable delays are about 15-30 ms without echo cancellers and 150 ms in international speech connections with echo cancellers. On IP world latency or delay limits are different. An acceptable latency is 300 ms and 450 ms latency is unacceptable. Delays appear from voice compressing and decompressing, voice packaging to ATM cell, different buffers on equipment through path of voice, Quality of Service (QoS) queues, switching of ATM cells in the network, build-out delay for accommodating Packet Delay Variation (PDV) and cells transmission time in physical network. Hybrid networks cause Echo where are mixed 2-wire cables and 4-wire cables. Acoustical feedback at the end user's terminal is on other phenomenon that causes echo. When speech channels are carried in ATM cells then echo cancellers are required at the interface of each speech circuit into the ATM network. [7]

2.2.2 Control plane

The control plane or signaling depends on what is ATM network structure and which AAL layer has been used to carry voice stream. The narrowband system and the ATM desktops use AAL1 layer to transfer voice stream over ATM networks.

Figure 2: Signaling on AAL1 layer [8]

The call handling signal has been mapped first on the narrowband site to Digital Subscriber Signaling System No. 1 (DSS1) or to the Private Integrated services Signaling System No. 1 (PSS1) signal through data link layer of ISDN (Q.921). The Signaling protocols between two IWFs can be DSS1, PSS1, or Channel Associated Signaling (CAS), but it must be the same protocol that has been used between the narrowband ISDN (N-ISDN) and the IWF. A Signaling ATM Adaptation Layer (SAAL) adapts different 64 kbps channels to cell of ATM in separate VCC. Voice over ATM for the desktop [4] is using B-ISDN and signal has mapped to a Digital Subscriber Signaling System No. 2 (DSS2) after SAAL adaptation. The DSS2 been mapping protocol has been specified in ITU-T specification named Q.2931.

Figure 3: End-to-end call using SS7 [5]

It is normal to use an existing Signalling System No. 7 (SS7) network to separate it from a transport layer. Between IWFs networks SS7 signaling is adapted to AAL2 Virtual Circuits (VC), but SS7 signaling is adapted to VC of AAL5 between IWF and PSTN. SS7 signaling is used also in call control to forward a call request to the Service and Transport Unit (STU) and to terminate an ISDN user part (ISUP) Signaling call request with the far end. Signaling network and transport network have different path and that is one reason to different delays on these two networks. The
call control delay is called post dial delay or post selection delay. The other delay is an ATM trunking bearer transit delay.

2.3 ATM services and descriptors

ATM services have two different main descriptors, namely traffic descriptors and QoS descriptors. Traffic descriptors ensure customer’s traffic integrity is Peak Cell Rate (PCR), Sustainable Cell Rate (SCR), Maximum Burst Size (MBS), Minimum Cell Rate (MCR), and sometimes Cell Delay Variation Tolerance (CDVT).

- Peak Cell Rate, defines the maximum cell rate over connections that source can submit

- Sustainable Cell Rate, defines average cell rate upper limit to connections without traffic violation

- Maximum Burst Size, defines maximum number of cells that can be sent back-to-back at the peak rate

- Minimum Cell Rate, parameter that is set by the MCR commitment request

- Cell Delay Variation Tolerance, defines the maximum cell delay variance

The QoS descriptors define QoS to guarantee network connection. These descriptors are Maximum Cell Transfer Delay (maxCTD), Cell Delay Variation (CDV), and Cell Loss Ratio (CLR).

- Maximum Cell Transfer Delay, CTD is a delay that is generated when cell is transferred from start point to end point on the network and maxCTD is an upper limit on CDT.

- Cell Delay Variation, defines maximum cell transfer delay time difference in the network. CDV is the maxCTD (worst case) minus fixed delay (best case).

- Cell Loss Ratio, defines ratio to the ratio of lost cells to total cells transmitted. Cell loss causes e.g. buffer overflow situation or wrong routing.

The ATM Forum has defined five different ATM service classes. Service provider and customer define beforehand what QoS class customer traffic is. With QoS classes teleoperator can balance or limit connection bandwidth to certain maximum value e.g. DS3 (Digital Service 3, 44.736 Mbps).

Figure 4: ATM service classes

- Constant Bit Rate services, bit rate that supports transport services that require rigorous timing controls and performance parameters. Supported devices or services are video, single voice channel, N*64kbps, DSn, En, Q.931 N-ISDN D-channel Signaling and circuit emulation. It has also a nickname “Continuous Bit Rate”.

- Variable Bit Rate services, bit rate that enables data traffic with average and peak traffic parameters. In future this class is divided to two separate services’, real-time-VBR (rt-VBR) service and non-real-time-VBR (nrt-VBR) service. Supported services are such as a bursty traffic like single voice channel with silence removal and compressed video packets.

- Available Bit Rate (ABR) services, designed for data applications for elastic applications such as a Client-Server applications, TCP/IP, and LAN applications plus Q.2931 ATM Signaling.

- Unspecified Bit Rate (UBR) services, bit rate that does not specify traffic related service guarantees. Used to fill bandwidth with data stream, provides “best-effort” service by the IP layer, and Q.2931 ATM Signaling.
3 Voice over ADSL

3.1 Standards

VoDSL and technologies that are related to the Asymmetric Digital Subscriber Line (ADSL) have been standardised by the several organisations. Physical layer standards have been done in American National Standards Institute (ANSI), in European Telecommunication Standards Institute (ETSI) and in the International Telecommunication Union (ITU). ATM standards have been done in ITU and in the ATM Forum and Internet Engineering Task Force (IETF) has done IP standards. ADSL Forum has its’ responsible area on ADSL architectures and share marketing information. There are several working groups, one of, which is VoDSL group whose aim is to prepare architectures, requirements, and recommendations. VoDSL sub-working groups are Broadband Loop Emulation Service (BLES) group, Voice with Multiservice Data Networks (VMSDN) group, and Technical Marketing (VTM) group. The BLES definition includes Plain Old Telephony Server (POTS), GR-303 interfaces on the Class-5 voice switch, Permanent Virtual Circuit (PVC) and AAL2. VoDSL equipment is management over network with the Simple Network Management Protocol (SNMP, RFC 1157) or with the Open Systems Interconnection (OSI) Common Management Information Protocol (CMIP, RFC 1095).

3.2 Digital Subscriber Line (DSL)

Two different working groups, the ADSL Forum and the Universal ADSL Working Group (UAWG), have originally developed DSL. DSL is a broadband technology over copper telephone lines. XDSL refers to the family of DSL technologies and the most popular are ADSL and High-bit-rate DSL (HDSL). The ANSI specification for ADSL uses 256 frequency channels for downstream transmission and 32 channels for upstream and channels’ bandwidth is 4,3125 kHz. ADSL technology provides asymmetrical data capacities downstream to customer’s site from 1.5 Mbps to 9 Mbps and upstream from customer’s site from 64 kbps to 1.5 Mbps. Typically in DSL technology the data part is continuously connected to the network but voice connection will be created when it is necessary. With DSL technology operator ordinary used high data transmission frequency through twisted pair and therefore there is 12000 feet limit to maximum line length.

VoDSL technology provides user voice traffic over telecommunications network and data applications over copper lines. Main target to VoDSL is packetized voice through packet networks. The amounts of ADSL voice lines are dependent on the line speed of ADSL and on the type and amount of compression. If line speeds are 384 kbps then maximum lines are 6 without compression and 40 with compression. Whereas line speed is 1.5 Mbps then maximum lines are 25 without compression and 150 with compression.

VoDSL technology is a platform that includes different equipment to server voice or data to customer over Public Switched Telephony Network (PSTN) or over packet backbone network. Equipment or Network Elements (NE) are customer equipment (e.g. telephones, fax, and modem), Integrated Access Device (IAD) like ADSL modem, Digital Subscriber Line Access Multiplexer (DSLAM) and voice gateway. Two basic components are a voice gateway (GW) that filters voice traffic from data network for sending to the PSTN and IAD that provides interface to the voice and data traffic between the customer equipment and the DSL network.
DSLAM is a switch that terminates DSL lines and routing voice and data to the appropriate network. DSLAM sends data and voice through access data switch to the packet backbone network or to the voice GW. Voice GW packetized voice to standard format to send to a Class 5 Switch. Class 5 Switch provides different voice services like dial tone, call routing and billing records. Physical transport layer can be Frame Relay (FR), ATM, or IP. When teleoperator packetized VoDSL traffic to different physical layers, then VoDSL technology comes near to other technology as Voice over Frame Relay (VoFR), Voice and Media over ATM (VMoA), or VoIP. Each technology has its own strengths, FR is common used technology, ATM has useful QoS traffic classifies and IP is future technology. AAL1 and AAL2 are both used as ATM adaptation layers, but AAL2 offers better traffic classification and efficient bandwidth use.

VoDSL signalling supports on mandatory the establishment of incoming calls and outgoing calls, the associated bearer channel’s allocation, recovery for disruptions of data flow. VoDSL technology is an end user technology and ATM networks are commonly used under DSL to offer faster transport layer. Signalling data is using analogy Signaling system (D channel) and system is carried voice and Signaling over the same path. System uses ATM networks to transport data and voice. There are two different form to offer signaling on VoDSL, CAS and Command Channel Signalling (CCS). The CAS is sent in the same AAL2 channel as the voice, but CCS is sent independent in own channel on the VCC. Some cases Signaling of VoDSL must be compatible to the telephony signal that can be SS7 ISUP, ISDN primary Rate Interface (ISDN-PRI), Q-reference point Signaling (QSIG), Digital Private Network Signaling System (DPNSS) and R2.

4 Voice over Cable TV

4.1 Standards

Main standardisation organisations are Multimedia Cable Network System (MCNS) partners, ETSI, and ITU. They all have created different standards of cable modem and these standards are used in different geographical and market areas. MCNS partners were cable operators and media companies from North America. For this reason MCNS cable standard named Data over Cable Service Interface Specification (DOCSIS) is used in North America area [10]. ETSI has standardised Digital Video Broadcasting (DVB) and document’s number is ETS 300 800 [11]. Other standard that is used in European area is Digital Audio Video Council (DAVIC) and these two standards dominate markets all over the world. MCNS and DVB/DAVIC standards are used in Asian market and South American markets use DVB/DAVIC standards. Both DVB/DAVIC and MCNS standards are nowadays ITU standards.

4.1.1 DVB/DAVIC standard

DAVIC is an association that has been established in 1994 [12]. DAVIC association has memberships that represent all sectors: manufacturing, service, research organisations, and governments. DAVIC has created the industry standard named DAVIC 1.4 Specification. Specification includes specification areas from architecture, interfaces, protocol layers, security, and interoperability. DVB standard was accepted by ETSI organisation in 1997. DVB specification includes specifications from audio, conditional access, interactivity, interfacing, measurement, Multimedia Home Platform (MHP), multiplexing, subtitling, and transmission. DAVIC and ETSI have worked together and nowadays these two DVB specification and DAVIC specification are identical.

DVB/DAVIC standards have been created to European markets. European Union (EU) has drawn up a directive that digital TV transmission must be based on DVB specification and EU has recommended to use DVB/DAVIC standards on digital audio and video services. The European Cable Communications Association (ECCA) recommends using these standards on equipment and cable operators demand that Cable Modems (CM) are compatible to standards. The DVB organisation has developed two different standards to audio, video, and data services. These two standards are DVB over cable (DVB-C) and DVB Return Channels for Cable and LMDS (DVB-RCC). DVB/DAVIC standards support different services, like digital TV, data communications and telephony. Physical layer is based on ATM technology with QoS.
The QoS is the main requirement in TCP/IP traffic. Users can choose only one or two equipment to home and use these services with Personal Computer (PC) or Set-Top-box (STB) through same infrastructure on cable network. STB equipment is used with interactive TV applications and with Internet applications. The CM system, where maybe have been integrated both LMDS and Hybrid Fiber Coax (HFC) technologies, is used with video, data, and voice applications.

4.1.2 MCNS/DOCSIS standard

MCNS standards were written first by North American multi cable service operators (MSO) and media companies. The DOCSIS specification of MCNS was developed for the transmission of data over cable network in the beginning of 1998 and it is based upon North American TV standards. First version of this standard didn’t support QoS, and this service was important to the voice service, but now this important service, QoS, has been added to the specification. The DOCSIS 1.1 specification includes fragmentation in the upstream and downstream, support QoS, and tiered services. The DOCSIS specification is for US market and for European market there is an EuroDOCSIS specification, which is similar to the DVB specification.

4.1.3 Other standards

There are some other organisations or working groups who have defined different specifications to the cable networks. The IPCDN (IP over Cable Data Network) develops and standardises SNMP MIBs for IP-capable data-over-cable systems. ATM Forum has been defined VoA and PacketCable has been developed interim specifications for interoperable interfaces.

4.1.4 DVB/DAVIC versus MCNS/DOCSIS

These two standards, DVB/DAVIC and MCNS/DOCSIS dominate markets all over the world and standards are developed for the same purpose of delivering bi-directional IP traffic to voice and video to customers’ terminal equipment over cable network. Both standards define devices STB and CM that can handle simultaneous video stream and data stream. The DAVIC specification has defined an out of band (OOB) channel to handle these different streams.

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<tr>
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<th>DVB/DAVIC</th>
<th>DOCSIS</th>
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<tr>
<td><strong>Downstream Modulation</strong></td>
<td>QPSK 16, 64</td>
<td>QPSK 16 QAM</td>
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<td>256 QAM (MPEG-2 transport)</td>
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<tr>
<td><strong>Upstream Modulation</strong></td>
<td>QPSK</td>
<td>QPSK 16 QAM</td>
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<tr>
<td><strong>Downstream Spectral Efficiency</strong></td>
<td>1.544 and 3.088 Mb/s</td>
<td>MPEG-2 transport: RS(128,122);</td>
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<td>1 MHz and 2 MHz bandwidth ATM cell protected with RS(55, 53)</td>
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<td>3333 and 6666 ATM cells</td>
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<td>MPEG-2 transport: RS(204,188)</td>
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<tr>
<td><strong>Upstream Spectral Efficiency</strong></td>
<td>256 kb/s… 6.176 Mb/s</td>
<td>160, 320, 640, 1280, and 2560 Kilobaud in channels</td>
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<tr>
<td></td>
<td>200 Hz … 4 MHz bandwidth ATM cell protected with RS(59, 53)</td>
<td>200 kHz, 400 kHz, 800 kHz, 1.6 MHz and 3.200 MHz bandwidth RS code</td>
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<tr>
<td></td>
<td>12000 ATM cells / s at 6.178 Mb/s</td>
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<tr>
<td><strong>Physical layer</strong></td>
<td>ATM framing (IPoA) AAL5</td>
<td>Ethernet Frames LLC/SNAP encapsulation MCNS</td>
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Table 1: Overview of DVB/DAVIC versus DOCSIS

4.2 Cable TV Architecture

Traditional CATV was a one-way transmission network where the system sent programs to a downstream direction. Nowadays CATV provides fast data transmission using cable modems, modems that are used to connect end user to the CATV network. A cable modem speed is from 3 Mbps to 50 Mbps and a line distance can be over 100 km. A CATV network structure is point-to-point and topology is a tree model and it uses broadcasting technology. CATV is using coaxial cables or HFC on networks because of high frequencies of transmission. On the network there are cable converters near the subscriber and the last kilometres to the customers site are usually coax cables.
The CATV network is nowadays a two-way traffic and it can use different services. These services have been classified to two classes, distributive services and interactive services. Distributive services are traditionally broadcasting streams to customers like television, radio and teletext. Interactive services have a two-transmission path, downstream from service centre towards customer and upstream from customers towards service centre. After telecommunication services deregulation CATV operators have been able to offer interactive services and telecommunication services. Cable operators offer telecommunication services using Internet Protocol (IP) based telephony system and Internet access services like web pages download. End User needs splitter with HPF for traditional services and the Cable Modem for computer based services. The HPF component allows only a high frequency stream like TV-channels to pass and filter all low frequency streams from both directories, from upstream and from downstream. When two Cable Modems send and receive traffic together then the Cable Modem Termination System (CMTS) must be used, because CMs can not send and receive messages to/from CMTS equipment. One CMTS can handle up to 2000 CMs on a single TV channel.

Both NEs use the same modulation methods; same frequency, same bandwidth, and it transmit rate. Upstream demodulator in CMTS equipment and modulator in CM equipment use Quadrature Phase-Shift Keying (QPSK) and 16-Quadrature Amplitude Modulation (QAM), frequency is between 5 to 65 MHz, bandwidth is e.g. 2 MHz, and traffic rate is 3 Mbps. To other directory, downstream, modulator in CMTS equipment and demodulator in CM equipment use 64-QAM and 256-QAM modulation, frequency is between 65 to 850 MHz, bandwidth is from 6MHz to 8 MHz, and traffic rate is between 27 to 56 Mbps. An amplitude and phase coding methods are used on QAM modulation and QPSK modulation method uses four phase states to a code two digital per phase shift.

Three CM configurations are an external CM, an internal CM, and an interactive set-top box. The external CM and internal CM are normal Network Interface Cards (NICs). External CM connects to the computer through an Ethernet connection or through Universal Serial Bus (USB). Externals CM can be used like Medium Access Control (MAC) layer bridge or like as router. The internal CM is an extra card that will be installed inside PC normally to a Peripheral Component Interconnect (PCI) bus. The interactive STB provides more TV channels on the same frequencies. There is a return channel that enables to use POTS for web browsing and for email.
4.3 PacketCable Architecture

The PacketCable is a trademark of Cable Television Laboratories Inc (CableLabs) and PacketCable includes different specifications and reports. These specifications defined requirements for Signaling, QoS, codecs, client provisioning, billing event message collections, PSTN interconnection, and security. The PacketCable specifications can be used packet-based voice, video and other high-speed multimedia services over CATV network. The PacketCable system is based on the DOCSIS 1.1 protocols and it offers IP-based voice services over other IP-based networks. [14]

Figure 8: PacketCable System Architecture [15]

The PacketCable system has general assumptions about the network architecture. These assumptions are that the architecture should be support many different service or rules

- an international infrastructure
- trust relationships or trust models
- toll-bypass over PSTN - PacketCable - PSTN networks are not allowed
- no PacketCaple specific elements is defined to a transit IP network
- support common VoIP services
- routing applications for learning the IP addresses
- support media and signaling transfer through the network
- an information data protection from other zones.

The PacketCable networks contain different NEs. A Media Terminal Adapters (MTA) contains the interface to a voice device e.g. the telephony and interface to network. Other functions that MTA provides are codecs, signaling, and encapsulation functions for VoIP, and QoS signaling. The CMTS is a head-end device and it offers Radio Frequency Interface (RFI) MAC protocol. CMs are connected over an HFC networks to the CMTS. A Call Management Server (CMS) controls the audio connections and it is called a Call Agent. There are different gateways between IP backbone transport network and PSTN network. The GWs are Media Gateway Controller (MGC), Media Gateway (MG), and Signaling Gateway (SG). The PacketCable networks regional are wide and technology is IP-based, so networks contain proxies that separate different domains or realms. Security features are offered with a kerberos technology. The Kerberos Key Distribution Center (KDC) gives tickets and ensures users’ authentication and authorisation.

4.4 CMS and SIP

The signaling and data paths are different between packet networks. Network includes different border routers, the border proxies for signaling and border gateways for media. The PacketCable supports two different signaling system, CMS-CMS and CMS-MGW signaling. These signaling systems have been defined in the Call Management Server Signaling (CMSS) protocol specification. The CMSS protocol is based on the SIP specification version 2.0. CMSS call signaling routing should be based to static routing tables. The
CMSS has same performance requirements as other telephone networks. Low end-to-end packet delay is limited to the same 300 ms value than the ITU recommendation. Packet loss must be low and this value is related to quality. User must receive the response immediately after the last digit from network and this delay is named “short post-dial delay”. The short post-pickup delay is the delay between a user picking up a ringing phone and the initial first talk in telephone.

CMS’s service-specific control functions support the telephony services; signals authentication, session by session based service authorisation, names and E.164 numbers translation, IP address based call routing, service-specific admission controls like policies and emergency calls, and signaling and service feature support. CMSS trust model means system that ensures reliable interfaces between single domain of cable network and various systems outside this inter-domain area. This domain has different GWs or proxies that have been connected to third party service providers and the PSTN networks. The CMTS servers offer signaling and data path to MTAs and to customers' equipment.

The CMSS of PacketCable is based on SIP protocol, but there are several SIP Extensions (SIP+) that supports basic telephony and custom calling features. A Distributed Call Signaling (DCS) is a term that identifies these SIP+ extensions. Servers that respond to call and in particular accept session request, a clients initiate requests and in particular initiate sessions, and a user agents. A reservation scheme extension allows network resources to be reserved earlier user alerting and network resource commitments. One extension is user privacy, that subscribers can make connections without identifying and without location information. The proxies can store state information in the entities and this feature reduces the storage requirements and reliability requirements. A media authorisation mechanism allows control access network resources equitably. The SIP+ includes mechanism that allows service provider functions e.g. billing and security features. The SIP+ extensions are also a provisional response to a SIP request and request sending to other user agents to initiate a new session.

5 Conclusions
These three technologies, VoATM, VoDSL, and VoCATV, offer new transfer methods to carry voice traffic. The TDM technology has its own strengths, but circuit based networks are inefficient when thus are compared to packet based networks. ATM networks are widely used in North American and ATM is based on cell technology. Main strength of ATM is QoS service classes. QoS of ATM and IP technology’s Type of Service (TOS) guarantee certain traffic class level to customers without significant delays or latencies.

The DSL technology is one noteworthy modem technology to consumers after analogy modems or ISDN. DSL offer better transfer speed than older modems and users can use VoDSL phones over IP networks. DSL uses copper lines and these lines between DSLAM and IAD limit line length to a few kilometres. Telecommunications Industry Analysis company RHK has predicted that DSL user amount in the USA will increase from the current 2.3 million users to the 17 million by 2004.

The CATV is an old technology, but new CATV technologies offer two different services, distributive services and interactive services. The VoCATV is included to interactive services. The CATV standardisation work is ongoing and two standards, DVB/DAVIC from European and MCNS/DOCSIS from North American, dominated other specifications. The CATV offers competitive way to use IP Telephone and customers can connect to the Internet via STBs or PCs. Telecommunications Industry Analysis company RHK has predicted that Internet access via CMs will increase in the USA from the current 4.9 million users to 18 million by 2004.

The VoA is lower layer technology where user traffic has been multiplexed to ATM cells and ATM cells send over telephone networks. VoDSL and VoCATV are modem technologies that are offered to consumers or to small business by operators. There is a pure VoA solution where ATM card has been installed to PC and this solution is powerful to use. Normally the transfer speed of these cards is high e.g. 25 Mbps and cards are used AAL5 layers. Problem is that card must be connected to fibre cable and this kind cable system is not so common and it is expensive if system is compared to other voice over packet network solutions.

These three “voice over” technologies that have been studied in this paper are not pure competitors to each other. VoA is used in core networks to offer transport mechanism to other voice technologies and use of ATM NIC cards is focused to certain areas e.g. call center environments and telemedicine. Only VoCATV and VoDSL technologies are quit a similar and they are competitors, but also these technologies have their own segments and own clientele.
### Acronyms:

- **AAL**: ATM Adaptation Layer
- **ABR**: Available Bit Rate
- **ADSL**: Asymmetric Digital Subscriber Line
- **ANSI**: American National Standards Institute
- **ATM**: Asynchronous Transfer Mode
- **B-ISDN**: Broadband ISDN
- **CATV**: Community Antenna Television
- **CBR**: Constant Bit Rate
- **CM**: Cable Modem
- **CMTS**: Cable Modem Termination System
- **DAVIC**: Digital Audio Video Council
- **DOCSIS**: Data over Cable Service Interface Specification
- **DSS**: Digital Subscriber Signaling System
- **DVB**: Digital Video Broadcasting
- **ETSI**: European Telecommunication Standards Institute
- **FR**: Frame Relay
- **GW**: Gateway
- **HFC**: Hybrid Fiber Coax
- **HPF**: High-pass-Filter
- **IETF**: Internet Engineering Task Force
- **ISDN**: Integrated Services Digital Network
- **ITU**: International Telecommunication Union
- **IWF**: Interworking Functions
- **LPF**: Low-pass-Filter
- **MCNS**: Multimedia Cable Network System
- **NID**: Network Interface Device
- **N-ISDN**: Narrowband ISDN
- **PDH**: Plesiochronous Digital Hierarchy
- **POTS**: Plain Old Telephone Service
- **PSS**: Private Integrated services Signaling System
- **PSTN**: Public Switched Telephony Network
- **QoS**: Quality of Service
- **SAAL**: Signaling ATM Adaptation Layer
- **SDH**: Synchronous Digital Hierarchy
- **SIP**: Session Initiation Protocol
- **SIP+**: SIP Extensions
- **SS7**: Signalling System No. 7
- **STB**: Set-Top-box
- **TDM**: Time Division Multiplexing
- **UBR**: Unspecified Bit Rate
- **VBR**: Variable Bit Rate
- **VC**: Virtual Circuits
- **VMoA**: Voice and Media over ATM
- **VoA**: Voice over ATM
- **VoCATV**: Voice over Cable Television
- **VoDSL**: Voice over Digital Subscriber Line
- **VoFR**: Voice over Frame Relay
- **VoIP**: Voice over IP
- **VTOA**: Voice and Telephony Services over ATM

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