ASL
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Voice over IP Public Address Software

The ASL PMC (Portable Media Carrier) software modules are part of the VIPA library of IP-enabled software solutions and provide multicast VoIP functionality on either Windows or Linux platforms.

The modules can be installed on any ASL VIPA operating platform, and are standard in the product software for the VIPET IP Audio Controller, the iPA400 and iPAM400 Amplifier Mainframes, the DVA01 PC/DVA Control and Display System, and the MCS02 Mini Control System. The modules use standard IPV4 UDP multicast with IGMPv3. This is compatible with all modern networks and operating systems, and the modules can also be run on any appropriately specified standard PC platform.

The PMC software stack enables efficient broadcast network audio using a variety of audio codecs. It is particularly suitable for deployments where:

(a) the same audio is being transmitted to a number of remote locations; or

(b) network bandwidth is a constrained resource; or

(c) synchronisation is required between output nodes because of overlapping audio zones.

The software stack is provided in two forms, dependent on the operating platform:

- Windows platform: standard Windows installable service
- VIPA operating platform/Linux platform: Linux Standard Base compliant daemon

The PMC software modules do not have a graphical user interface. Its control API is exposed using ASL's JCOP protocol for network-transparent RPC.

The PMC modules are fully interoperable with ASL's VIPA software library.

The PMC module is standard in ASL's Audio Server module. The Audio server module acts as an abstraction layer for the platform's audio hardware, and is the means by which other VIPA modules and third-party integrators access that hardware.

The Audio Server presents one or more PMC broadcasting channels as if they were physical soundcard outputs. Similarly, the Audio Server can create dynamic PMC listening channels, attach them to a given PMC broadcast channel, and present them as if they were physical soundcard inputs.

When this is combined with the internal I/O routing within the Audio Server, the integrator is able to construct a networked audio channel (to one or more remote nodes) that is transparent to the top-level client. For example, client software may make an audio announcement to zones 1, 2 and 3, without being aware that zone 3 is in fact a remote zone being addressed over an IP link.

Application
- IP-based PA (Public Address)
- IP-based LLPA (Long Line Public Address)

Feature List
- Real-time digital audio distribution via Ethernet
- No overall limit on network channel capacity
- Supports switched and repeater fast Ethernet networks
- Compatible with fibre optic and gigabit Ethernet as well as Networks running over legacy copper
- Ethernet infrastructure can be used simultaneously for audio and data communications
- High-precision time synchronisation between nodes
- Professional 48 kHz 16-bit audio as standard with options for various audio compression codecs
- Adaptive latency monitoring for synchronised audio between network nodes
- Flexible network audio routing capabilities
- Remote management via rich JCOP API
- Non-volatile storage of configuration parameters
- Software module upgradeable over Ethernet connection

Platform Requirements

Either
- VIPA operating platform

Or
- Standard PC with 256 MB RAM and 800 MHz Pentium-class CPU (or better)
- Windows operating system (Windows 2000 SP4 or later) or Linux operating system (Kernel 2.6 or later)
- 100 MB free disk space
- Sound card supporting ASIO (Windows platform) or ALSA (Linux platform)
- TCP/IP connectivity (network speed required depends on VoIP codec used)

In addition to the network-transparent audio PMC channels offers 3-channel sweepable EQ with level control.

PMC Protocol

The PMC protocol is a royalty-free specification for the transmission of streaming media over unicast or multicast UDP (User Datagram Protocol) where the time of final delivery needs to be controlled. It includes provision for inter-zone synchronisation and quality of service feedback to the audio source node.

Awareness of the details of the protocol is not necessary to use the PMC modules; those wishing to develop third-party supporting software may contact ASL for additional information.

Application				PMC	SNMP	JCOP
Transport				UDP		TCP
Network	RARP	ARP	ICMP	IP		
Logical Link	802.3 Interface					
Physical	Fast Ethernet Interface					

Network Bandwidth Dependencies

The Multicast broadcast technique that is used enables a live broadcast to be made to every location in a network with no more bandwidth requirement that for an announcement to a single location. The bandwidth required for the system increases with simultaneous but different broadcast requirements, rather than simultaneous identical broadcast requirements, i.e. with the number of 'all call' microphones rather than the number of broadcast destinations. The effective maximum number of remote locations is very large, but due to the use of Multicast is not limited for VoIP broadcasts by the network bandwidth. The use of Multicast also eliminates problems caused by 'necking points' in the network bandwidth, as long as the minimum bandwidth as detailed below is maintained throughout the network.

The bandwidth requirement detailed above is for a live VoIP broadcast. If centrally recorded DVA messages are to be distributed to the remote locations for scheduled automatic announcements, then these messages are sent individually to each location. The messages are then saved at, and individually played at, each location. This may use a higher bandwidth than the VoIP broadcasts, if this bandwidth is available, and is done 'in the background' in between live broadcasts. By default the distribution of recorded DVAs will proceed at the maximum rate that the network will stand, however if there are any requirements to limit the maximum bandwidth usage by the PA system then this message distribution can be capped to proceed at a lower maximum rate.

The system's data carrying capability also enables the implementation of a Text To Speech capability, and centrally defined Text To Speech messages can be distributed to the remote locations with very low bandwidth requirements, as the text only is sent, and the Text To Speech engines run at each remote location.

Audio Formats and Codecs

The PMC protocol is codec-agnostic and does not dictate a particular media format. The stream format is encoded in the packet headers, so no pre-negotiation is required between broadcaster and listener.

The choice of audio codec is governed by the desired trade-off between audio quality and network bandwidth requirements. The typical bandwidth requirements on networks supporting multicast operation are shown in the following table.

Format		Bandwidth per Channel (Multicast Operation on local network)		Network Latency[1]		Application
				Min	Max	
Uncompressed	48 kHz *(recommended)*	1 Mbit/s (*minimum*)	880 kbit/s for audio, plus margin for control	10-20 ms	85 ms (default)	Highest quality audio in LAN/high bandwidth WAN deployments
Compressed	Speex (robustified) (*recommended*)	160 kbit/s (*recommended*)	128 kbit/s for audio, plus margin for control	30 ms		General purpose voice and BGM (Background Music)
	Speex	64 kbit/s (*minimum*)	50 kbit/s for audio, plus margin for control	30 ms		Voice announcements

1) If the network does not support multicast operation, then the bandwidth at the originating point should be the figures shown above multiplied by the number of end points.
2) For maximum bandwidth savings from the use of multicast IP, ASL recommend a star topology network (active switches rather than passive hubs) and routers/switches that support layer 2 (MAC-based) multicast.
3) The TTL (Time-to-live) for outgoing packets defaults to 9 but is configurable, and the multicast group and ports used are completely configurable to use any legal address and port.
4) Latency introduced by a VIPA system for message transmission.
5) The maximum network latency depends on the buffering selected. In a standard VIPA system with block size=1024 and prefill=2048, the maximum latency is 4096 frames @ 85 ms.

   one-way latency=(2 x block size + prefill) ÷ 48000 (the uncompressed sample rate)

   For latency critical applications the block size and buffering can be reduced. For example, with block size=256 and prefill=512, the maximum latency will be 1024 frames @ 20 ms. This increases CPU load and thus reduces scope for additional processes on the unit.

   Therefore the minimum latency depends on the system configuration. For compressed audio the minimum latency also depends on the compression/decompression algorithm deployed in the codec.
6) In order to minimise interruptions to the audio stream, we recommend a network with a packet loss rate of no more than 0.01% and codecs robust to packet losses. The ASL system can however be ruggedised by using buffering and other techniques for operation on low quality networks, with a resultant increase in latency.
7) The number of simultaneous pre-recorded and live announcements is limited only by network bandwidth.
8) Connections across a shared or open network should use a dedicated VPN (Virtual Private Network) for secured data transfer.

For further details, and for information on other products, please visit www.asl-control.co.uk.

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