Case Study IP Telephony Deployment: Australian Catholic University

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Introduction

The Australian Academic and Research Network (AARNet) is a nationwide high−speed IP network that interconnects 37 Australian universities as well as the Commonwealth Scientific and Industrial Research Organization (CSIRO).

AARNet was initially built as a data network, but it has carried Voice over IP (VoIP) since early 2000. The VoIP network currently deployed is a toll−bypass solution that carries VoIP calls between the universities and the CSIRO private automatic branch exchanges (PABXs). It also provides public switched telephone network (PSTN) gateways that allow PSTN to hop off at the most cost−effective point. For example, a call from a PABX phone in Melbourne to a PSTN phone in Sydney is carried as VoIP from Melbourne to the Sydney PSTN gateway. It is there connected to the PSTN.

Australian Catholic University (ACU) is one of the universities that connects to AARNet. In late 2000, ACU began an IP Telephony deployment that deployed approximately 2,000 IP phones across six university campuses.

This case study covers the ACU IP Telephony deployment. The project is completed. However, there are significant architectural issues to address in the AARNet backbone if the network is to scale when other universities follow in the footsteps of ACU. This document describes these issues and proposes and discusses various solutions. The ACU IP Telephony deployment is likely to be adjusted later in order to fall in line with the final recommended architecture.

Note: Deakin University was the first Australian university to deploy IP Telephony. However, Deakin University does not use AARNet to carry IP Telephony traffic.

AARNet

The Australian universities and CSIRO built AARNet in 1990 through the Australian Vice−Chancellors' Committee (AVCC). Ninety−nine percent of Australian Internet traffic was to the founding members during the first few years. A small amount of commercial traffic was from organizations that had a close association with the tertiary and research sector. Use by the non−AARNet userbase increased to 20 percent of the total traffic by late 1994.

The AVCC sold the commercial customer base of AARNet to Telstra in July of 1995. This event spawned what was eventually to become Telstra BigPond. This stimulated further growth of the commercial and private use of the Internet in Australia. The transfer of intellectual property and expertise resulted in the development of the Internet in Australia. Otherwise, this would not have occurred at such a rapid rate.

The AVCC developed AARNet2 in early 1997. It was a further refinement of the Internet in Australia, which employs high−bandwidth ATM links and Internet services under a contract with Cable & Wireless Optus (CWO) Limited. The rapid deployment of IP services by CWO to meet the AARNet2 requirements was due in part to the transfer of knowledge and expertise from AARNet.

ACU

ACU is a public university that was established in 1991. The university has approximately 10,000 students and 1,000 staff. There are six campuses on the east coast of Australia. This table shows the ACU campuses and their locations:

Campus	City	State
Mount Saint Mary	Strathfield	New South Wales (NSW)
MacKillop	North Sydney	New South Wales (NSW)
Patrick	Melbourne	Victoria (VIC)
Aquinas	Ballarat	Victoria (VIC)
Signadou	Canberra	Australia Capital Territory (ACT)
McAuley	Brisbane	Queensland (QLD)

ACU relied on a Telstra Spectrum (Centrex) solution before the rollout of the IP Telephony solution that this case study describes. The move to IP Telephony was driven mainly by the desire to reduce cost.

CSIRO

CSIRO has approximately 6,500 staff at numerous sites in Australia. CSIRO conducts research in areas such as agriculture, minerals, energy, manufacturing, communications, construction, health, and the environment.

CSIRO was the first organization to use AARNet for VoIP. The organization pioneered the early work done in this area.

AARNet

The AARNet backbone is a significant component in any university IP Telephony deployment. It provides the interconnection of universities with two main services in the voice area:

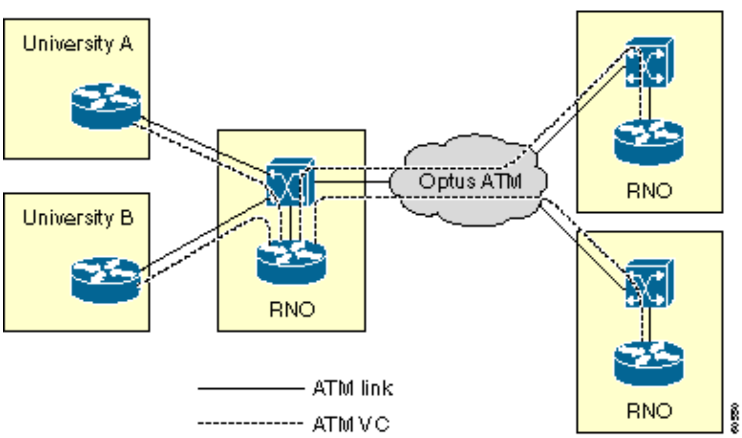
* Transport of VoIP Realtime Transport Protocol (RTP) packets with the guarantee of Quality of Service (QoS) appropriate to voice
* Low−cost hopoff point to the PSTNs around the country

This section describes the current AARNet architecture and how it delivers these services. It also outlines some of the scalability issues that arise as more universities deploy the IP Telephony solution. Finally, it discusses the possible solutions for these scalability issues.

AARNet Topology

AARNet consists of a single POP (point of presence) in each state. The POPs are referred to as Regional Network Operations (RNOs). Universities connect to the RNO in their respective state. The RNOs in turn are interconnected by a full mesh of Optus ATM PVCs. Together they constitute AARNet.

The typical RNO consists of one Cisco LS1010 ATM switch and one ATM−attached router. The RNO router connects to each university router by a single ATM PVC across an E3 microwave link. Every RNO router also has a full mesh of ATM PVCs that the Optus ATM network provides to all other RNOs. This diagram represents the general AARNet topology of the network:



There are numerous exceptions to the topology. Some of them are significant from a voice perspective. These are some exceptions:

* The RNO in Victoria uses classical IP over ATM (RFC 1577) instead of PVCs to connect the universities to the RNO.
* Rural universities typically connect back to the RNO by Frame Relay or ISDN.
* Some large universities have more than one link back to the RNO.

This table shows the states and territories that currently have an RNO. The table includes capital cities for readers who are not familiar with Australian geography.

State	Capital City	RNO?	Campus Connections
New South Wales	Sydney	Yes	TBD
Victoria	Melbourne	Yes	TBD
Queensland	Brisbane	Yes	TBD
South Australia	Adelaide	Yes	TBD
Western Australia	Perth	Yes	TBD

Australian Capital Territory	Canberra	Yes	TBD
Northern Territory	Darwin	No	−−
Tasmania	Hobart	No	−−

Quality of Service

Parts of AARNet are already QoS−enabled for voice as a result of the VoIP toll−bypass project. QoS is necessary for voice traffic in order to provide these features, which minimize delay and jitter and eliminate packet loss:

- Policing Mark down voice traffic from non−trusted sources.
- Queuing Voice must be given priority over all other traffic to minimize delay during link congestion.
- Link Fragmentation and Interleaving (LFI) Data packets must be fragmented and voice packets interleaved on slow links.

Traffic must be classified to properly police and queue voice packets. This section describes how classification is done on AARNet. Subsequent chapters describe the policing and queuing implementation.

Classification

Not all traffic gets the same QoS. Traffic is classified into these categories to selectively provide QoS:

- Data
- Voice from known and trusted sources
- Voice from unknown sources

Only trusted devices are given high−quality QoS on AARNet. These devices are mainly gateways identified by IP address. An access control list (ACL) is used to identify these trusted sources of voice.

```
access−list 20 permit 192.168.134.10
access−list 20 permit 192.168.255.255
```

IP precedence is used to distinguish voice traffic from data traffic. Voice has an IP precedence of 5.

```
class−map match−all VOICE
match ip precedence 5
```

Combine the previous examples to identify packets from a trusted source.

```
class−map match−all VOICE−GATEWAY
match class−map VOICE
match access−group 20
```

Use the same principles to identify voice packets from an unknown source.

```
class−map match−all VOICE−NOT−GATEWAY
match class−map VOICE
match not access−group 20
```

Policing

Voice traffic from a non−trusted source is classified and marked down when traffic arrives on an interface. These two examples show how policing is performed depending on what type of traffic is expected to arrive on a given interface:

The router looks for non−trusted voice packets and changes their IP precedence to 0 if there are trusted voice sources downstream.

```
policy−map INPUT−VOICE
class VOICE−NOT−GATEWAY
set ip precedence 0

interface FastEthernet2/0/0
description Downstream voice gateways
service−policy input INPUT−VOICE
```

The router looks for all voice packets and changes their IP precedence to 0 if there are no known voice sources downstream.

```
policy−map INPUT−DATA
class VOICE
set ip precedence 0

interface FastEthernet2/0/1
description No downstream voice gateways
service−policy input INPUT−DATA
```

Non−voice Queuing

All VoIP in AARNet was toll−bypass until recently. This condition results in relatively few VoIP endpoints. The current queuing design distinguishes between interfaces that have VoIP devices downstream and interfaces that do not. This section discusses queuing on non−VoIP interfaces.

A non−voice interface is configured for either weighted fair queuing (WFQ) or Weighted Random Early Detection (WRED). These can be configured directly on the interface. However, the queuing mechanism is applied by means of a policy map in order to make it easy to change the queuing mechanism on a given interface type. There is one policy map per interface type. This reflects the fact that not all queuing mechanisms are supported on all interfaces.

```
policy−map OUTPUT−DATA−ATM
class class−default
fair−queue

policy−map OUTPUT−DATA−VIP−ATM
class class−default
random−detect

policy−map OUTPUT−DATA−ETHERNET
class class−default
fair−queue

policy−map OUTPUT−DATA−VIP−ETHERNET
class class−default
random−detect

policy−map OUTPUT−DATA−SERIAL
class class−default
fair−queue

policy−map OUTPUT−DATA−VIP−SERIAL
class class−default
random−detect
```

The policy maps are attached to the respective interfaces and are specific to interface types. For example, this simplifies the process of changing the queuing mechanism on Versatile Interface Processor−based (VIP−based) Ethernet ports from WRED to WFQ. It requires a single change in the policy map. The changes are made to all VIP−based Ethernet interfaces.

```
interface ATM0/0
service−policy output OUTPUT−DATA−ATM

interface ATM1/0/0
service−policy output OUTPUT−DATA−VIP−ATM

interface Ethernet2/0
service−policy output OUTPUT−DATA−ETHERNET

interface Ethernet3/0/0
service−policy output OUTPUT−DATA−VIP−ETHERNET

interface Serial4/0
service−policy output OUTPUT−DATA−SERIAL

interface Serial5/0/0
service−policy output OUTPUT−DATA−VIP−SERIAL
```

Low Latency Queuing

Any interface that has downstream−trusted VoIP devices is configured for Low Latency Queuing (LLQ). Any packet that makes it through the incoming interface classification and retains a precedence of 5 is subject to LLQ. Any other packet is subject to either WFQ or WRED. This depends on the interface type.

Separate policy maps are created for each interface type in order to make QoS easier to administer. This is similar to the non−voice queuing design. However, multiple policy maps exist for each interface type. This is because the capacity of the interface types for carrying voice traffic varies depending on link speed, PVC settings, and so on. The number in the policy map name reflects the number of calls catered for 30 calls, 60 calls, and so on.

```
policy−map OUTPUT−VOICE−VIP−ATM−30
class VOICE
priority 816
class class−default
random−detect

policy−map OUTPUT−VOICE−VIP−ATM−60
class VOICE
priority 1632
class class−default
random−detect

policy−map OUTPUT−VOICE−ATM−30
class VOICE
priority 816
class class−default
random−detect

policy−map OUTPUT−VOICE−ATM−60
class VOICE
priority 1632
class class−default
random−detect

policy−map OUTPUT−VOICE−ETHERNET−30
class VOICE
priority 912
```

```
class class−default
fair−queue

policy−map OUTPUT−VOICE−VIP−ETHERNET−30
class VOICE
priority
class class−default
random−detect

policy−map OUTPUT−VOICE−HDLC−30
class VOICE
priority 768
class class−default
fair−queue
```

The policy maps are attached to the respective interfaces. In this example, the policy map is specific to an interface type. Currently no special treatment is given to voice signaling. The policy maps can easily be amended in one place if this becomes a requirement at a later stage on a given interface type. The change takes affect for all interfaces of that type.

```
Interface ATM0/0
service−policy output OUTPUT−VOICE−ATM−30

interface ATM1/0/0
service−policy output OUTPUT−VOICE−VIP−ATM−30

interface Ethernet2/0
service−policy output OUTPUT−VOICE−ETHERNET−60

interface Ethernet3/0/0
service−policy output OUTPUT−VOICE−VIP−ETHERNET−60

interface Serial4/0
service−policy output OUTPUT−VOICE−SERIAL−30

interface Serial5/0/0
service−policy output OUTPUT−VOICE−VIP−SERIAL−60
```

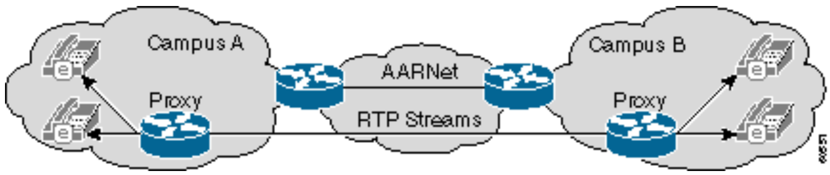
LLQ Scalability

The queuing mechanism has some scalability issues. The main issue is that it relies on knowing the IP address of every trusted VoIP device in the network. This was a reasonable limitation in the past when there was a limited number of VoIP gateways handling toll−bypass. The number of VoIP endpoints dramatically increases, and it becomes more and more impractical with the deployment of IP Telephony. The ACLs become too long and too hard to manage.

The ACLs have been appended to trust traffic from a specific voice IP subnetwork at each ACU campus in the case of ACU. This is an interim solution. These longer−term solutions are being investigated:

- H.323 proxy
- QoS ingress policing

The main idea behind the H.323 proxy solution is to have all RTP traffic enter AARNet from a given campus by means of a proxy. AARNet sees all RTP traffic from a given campus with a single IP address, as this diagram shows:

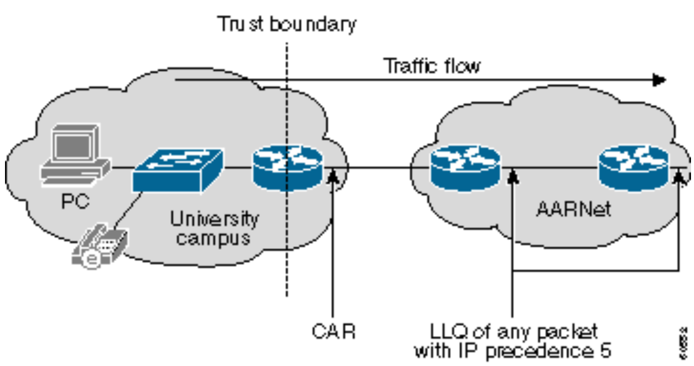


The number of entries in the QoS ACLs is limited to one line per campus if this scheme is deployed consistently. This scheme still has the potential to add up to 100 or more entries since there are 37 universities with multiple campuses. This too is not scalable. It might be necessary to move to a design with a single or limited number of shared super−proxies at each RNO. This reduces the number of trusted IP addresses to six. However, this opens up a QoS policing issue on the path from the campus to the proxy at the RNO.

Note: Cisco CallManager intercluster trunks do not currently work through an H.323 proxy because the intercluster signaling is not native H.225.

QoS ingress policing is an alternative solution. A trust boundary is established at the point where the campus connects to the RNO with this design. Traffic that enters AARNet is policed by the Cisco IOS® Committed Access Rate (CAR) feature at this boundary. A university that uses AARNet for VoIP subscribes to a certain amount of AARNet QoS bandwidth. CAR then monitors traffic that enters AARNet. Excess traffic has IP precedence marked down to 0 if the amount of RTP traffic with IP precedence 5 exceeds the subscribed bandwidth.

This diagram shows a CAR configuration:



This example shows how a CAR configuration handles this policing:

```
Interface a1/0.100
rate−limit input access−group 100 2400000 0 0 conform−action set−prec−transmit 5
exceed−action set−prec−transmit 0

access−list 100 permit udp any range 16384 32767 any range
 16384 32767 precedence critical
```

These are some advantages of a CAR configuration approach:

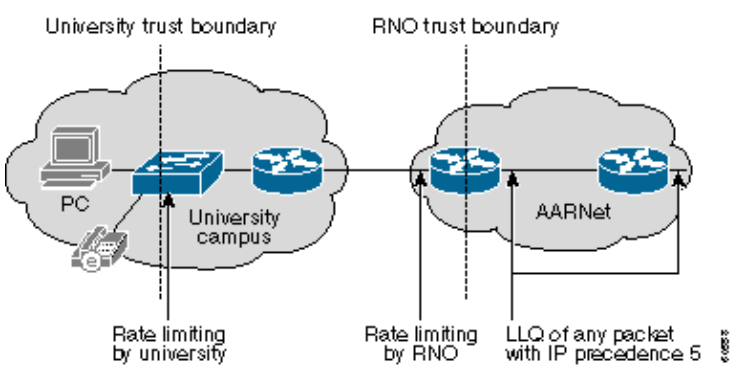
- The core no longer needs to handle policing. It is now handled at the trust boundary. Therefore, the LLQ in the core does not need to know about trusted IP addresses. Any packet with an IP precedence of 5 in the core can safely be subject to LLQ because it has already passed the policing at the ingress.
- No assumptions are made about the VoIP architecture, equipment, and protocols that individual universities choose. A university can choose to deploy a Session Initiation Protocol (SIP) or Media Gateway Control Protocol (MGCP) that does not work with H.323 proxies. VoIP packets receive the appropriate QoS in the core as long as they have an IP precedence of 5.
- CAR is resilient against QoS Denial of Service (DoS) attacks. A QoS DoS attack that originates from a university cannot damage the core. CAR limits the attack, which cannot generate more traffic than

what is present when the maximum number of allowed VoIP calls is active.

VoIP calls to or from that campus can suffer during an attack. However, it is up to the individual university to protect itself internally. The university can tighten the CAR ACLs on the router so that all but selected VoIP subnetworks have the IP precedence marked down.

Each campus has an internal trust boundary at the point where users connect to the campus LAN in the ultimate design. Traffic with an IP precedence of 5 that this trust boundary receives is limited to 160 kbps per switch port, or two G.711 VoIP calls. Traffic in excess of this rate is marked down. Implementation of this scheme requires Catalyst 6500 switches or something similar with rate limiting functionality.
- Bandwidth provisioning in the core simplifies as each university subscribes to a fixed amount of QoS bandwidth. This also makes QoS billing simple because each university can pay a flat monthly fee based on a QoS bandwidth subscription.

The main weakness in this design is that the trust boundary is located at the university router, so the universities must be able to correctly administer CAR. The trust boundary is pulled back into the RNO. RNO−administered equipment handles the policing in the ultimate design. This design requires hardware−based rate limiting such as the Catalyst 6000 switch or a Cisco 7200 Network Services Engine (Cisco 7200 NSE−1) processor. However, it gives AARNet and RNOs complete control over QoS policing. This diagram shows this design:



Link Fragmentation and Interleaving

VoIP is only being carried across relatively high−speed ATM virtual circuits (VCs). Therefore, no LFI is required. VoIP can also be transported across Frame Relay Forum (FRF) or leased lines to rural universities in the future. This requires LFI mechanisms such as Multilink PPP (MLP) with Interleave or FRF.12.

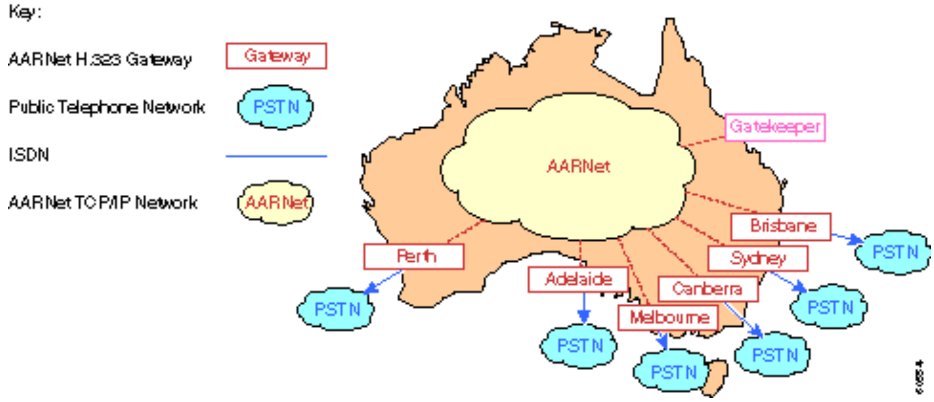
Gateways

There are two kinds of H.323 gateways in AARNet:

- PSTN PSTN to VoIP gateway
- PABX PABX to VoIP gateway

The distinction between a PSTN and PABX gateway is mainly functional. PSTN gateways provide connectivity to the PSTN. The PABX gateways connect a university PABX to the VoIP backbone. The same physical box acts as both a PSTN and a PABX gateway in many cases. There are currently 31 gateways in the ACU IP Telephony solution. Most of these gateways are Cisco AS5300 Universal Access Servers. The other gateways are Cisco 3600 series routers or Cisco 2600 series routers. A minimum of ten additional gateways are expected to be added during Q2CY01. AARNet carried approximately 145,000 VoIP calls in April of 2001.

AARNet has deployed PSTN−attached H.323 gateways in most major cities, as this diagram shows:



Universities can use these gateways to make outbound calls to the PSTN. Universities have to maintain their own trunks for inbound calls because they are not currently supported. AARNet can negotiate a very competitive price with the carrier because of the volume of calls that go through these gateways. Calls can also be dropped off at the most cost−effective point. For example, someone in Sydney who calls a Perth number can use the Perth gateway and only be charged for a local call. This is also known as Tail End Hop Off (TEHO).

A single gatekeeper is deployed to perform E.164 to IP address resolution. All calls to the PSTN are sent to the gatekeeper, which then returns the IP address of the most appropriate gateway. Refer to the Dial Plans and Gatekeeper sections for more detailed information on gatekeepers.

Billing and Accounting

The PSTN gateways use RADIUS and authentication, authorization, and accounting (AAA) for billing purposes. Each call through a gateway generates a Call Detail Record (CDR) for each call leg. These CDRs are posted to the RADIUS server. The IP address of the Cisco CallManager in the CDR uniquely identifies the university and ensures that the correct party is billed.

Gateway Security

Protecting the PSTN gateways against DoS attacks and fraud is a major concern. H.323 clients are widely available. Microsoft NetMeeting is bundled with Microsoft Windows 2000, so it is relatively easy for a non−technical user to place free calls through these gateways. Configure an inbound ACL that permits H.225 signaling from trusted IP addresses to protect these gateways. This approach has all the same scalability issues that the QoS section describes. The number of entries in the ACL grows as the number of trusted H.323 endpoints grows.

H.323 proxies offer some relief in this area. The gateway ACLs need to permit one IP address per university campus if all calls through the PSTN gateway pass through a campus proxy. Two IP addresses as a redundant proxy is desirable in most cases. Even with proxies, the ACL can contain more than 100 entries.

The proxy must be protected via ACLs since any H.323 can set up a call through the proxy. The proxy ACL must permit local H.323 devices as local policy requires since this is done on a per−campus basis.

The IP addresses of the two Cisco CallManagers must be included in the gateway ACLs if a campus wants to allow only calls from IP phones to use the AARNet PSTN gateways. The proxies do not add any value in this situation. The number of required ACL entries is two either way.

Note that intercampus IP phone−to−IP calls do not need to pass through the proxy.

Dial Plans

The current VoIP dial plan is straightforward. Users can place these two types of calls from a VoIP gateway perspective:

- Call a phone at a different campus but at the same university.
- Call a PSTN phone or a phone at a different university.

The gateway dial peers reflect the fact that there are only two types of calls. Basically there are two VoIP dial peer types, as this example shows:

```
dial−peer voice 1 voip
destination−pattern 7&
session−target ipv4:x.x.x.x

dial−peer voice 1 voip
destination−pattern 0&&&
session−target ras
```

The first dial peer is used if someone calls extension 7... at another campus in this example. This call is routed directly to the IP address of the remote gateway. Since the gatekeeper is bypassed, Call Admission Control (CAC) is not performed.

The second dial peer is used when the call is for a PSTN number. This can be either one of these items:

- The number of a phone in the PSTN
- The fully−qualified PSTN number of a phone at a different university

The call is sent to the gatekeeper by means of an admission request (ARQ) message in the first case. The gatekeeper returns the IP address of the best PSTN gateway in an admission confirm (ACF) message.

The call is also sent to the gatekeeper by means of an ARQ message in the second case. However, the gatekeeper returns an ACF message with the IP address of the VoIP gateway at the university that receives the call.

Gatekeeper

AARNet currently operates a single gatekeeper. The sole purpose of this gatekeeper is to perform call routing in the form of E.164 to IP address resolution. The gatekeeper does not perform CAC. The number of PABX trunks connected to the gateways limits the number of simultaneous calls. The core bandwidth caters for all trunks in use at once. This changes with the rollout of IP Telephony at ACU and other universities. There is no natural limit on the number of simultaneous VoIP calls that can be sourced in or out of a given campus in this new environment. The available QoS bandwidth can be oversubscribed if too many calls are initiated. All calls can suffer from poor quality under this condition. Use the gatekeeper to provide CAC.

The distributed nature and potential size of the university voice network lends itself to a distributed gatekeeper architecture. One possible solution is to have a two−tier hierarchical gatekeeper design in which each university maintains its own gatekeeper. This university gatekeeper is referred to as a tier 2 gatekeeper. AARNet operates a *directory* gatekeeper that is referred to as a tier 1 gatekeeper.

Universities must use this two−tier approach to use a gatekeeper for call routing between Cisco CallManager clusters. The gatekeeper routes calls based on a 4− or 5−digit extension in this scenario. Each university requires its own gatekeeper. This is because extension ranges overlap between universities since this is a locally−administered address space.

The university tier 2 gatekeepers perform CAC for calls to and from that university only. It also performs E.164 resolution for calls between only the campuses of that university. The call is routed by the tier 2 gatekeeper to the tier 1 gatekeeper by means of a location request (LRQ) message if someone calls an IP phone at another university or calls the PSTN through an AARNet gateway. The LRQ is forwarded to the tier 2 gatekeeper of that university if the call is for another university. This gatekeeper then returns an ACF message to the tier 2 gatekeeper at the university where the call originates. Both tier 2 gatekeepers perform CAC. They only proceed with the call if there is sufficient bandwidth available at both the calling and called zones.

AARNet can choose to treat the AARNet PSTN gateways like those of any university. Their own tier 2 gatekeeper looks after them. The tier 1 gatekeeper can also act as the tier 2 gatekeeper for these gateways if load and performance permit.

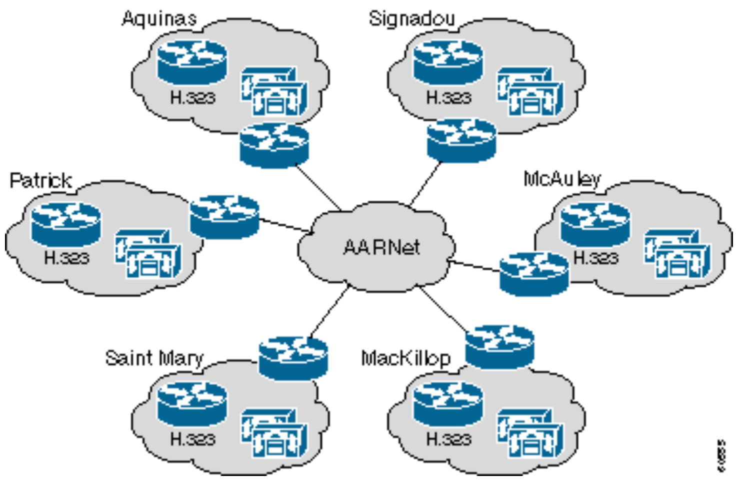
Each of the gatekeepers (including the AARNet directory gatekeeper) needs to be replicated because the gateways are such a critical component. Each university needs to have two gatekeepers. It is possible for Cisco IOS gateways to have alternate gatekeepers, as in the case of Cisco IOS Software Release 12.0(7)T. However, this is not currently supported by Cisco CallManager or any other third−party H.323 device. Do not use this feature at this time. Use a simple Hot Standby Router Protocol−based (HSRP−based) solution instead. This requires that both gatekeepers sit on the same IP subnetwork. HSRP determines which gatekeeper is active.

ACU IP Telephony Network

This table shows the approximate number of IP phones installed at the campuses of ACU:

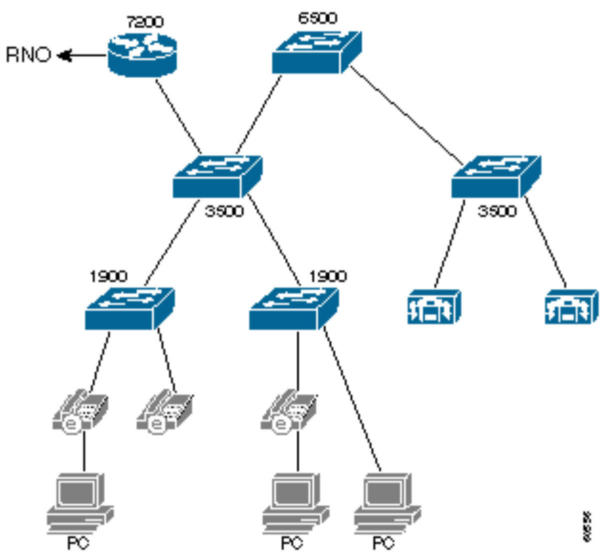
Campus	City	Approximate IP Phones
Mount Saint Mary	Strathfield	400
MacKillop	North Sydney	300
Patrick	Melbourne	400
Aquinas	Ballarat	100
Signadou	Canberra	100
McAuley	Brisbane	400
	Total:	***1700***

ACU recently deployed an IP Telephony solution. The solution consists of a cluster of two Cisco CallManagers, a Cisco 3640 gateway at each campus, and IP phones. AARNet interconnects the campuses. This diagram depicts the high−level topology and the various components of the ACU IP Telephony network:



ACU Network Topology

This diagram shows a typical ACU campus. Each campus has three layers of Catalyst switches. The wiring closet houses the older Catalyst 1900 switches. The Catalyst 1900 switches connect back to the Catalyst 3500XL switch by means of Extended Framing. These connect back to a single Catalyst 6509 switch by means of Gigabit Ethernet (GE). A single Cisco 7200 VXR router connects the campus to AARNet by an ATM VC to the local RNO.



The connectivity method to the RNO differs slightly from state to state, as this table shows. Victoria is based on Classical IP over ATM (RFC 1577). The other RNOs have a straight PVC setup with RFC 1483 encapsulation. Open Shortest Path First (OSPF) is the routing protocol used between ACU and the RNOs.

Campus	State	Connectivity to RNO	Routing Protocol
Mount Saint Mary	NSW	RFC 1483 PVC	OSPF
MacKillop	NSW	RFC 1483 PVC	OSPF
Patrick	VIC	RFC 1577 Classical IP over ATM	OSPF
Aquinas	VIC	RFC 1577 Classical IP over ATM	OSPF

Signadou	ACT	RFC 1483 PVC	OSPF
McAuley	QLD	RFC 1483 PVC	OSPF

The Catalyst 1900 series switches support trunking on the uplinks only. Therefore, the IP phones and PCs are all in one large VLAN. In fact, the entire campus is one large VLAN and broadcast domain. Secondary IP subnetworks are used because of the large number of devices. The IP phones are on one IP subnetwork, and the PCs are on another. The AARNet core trusts the IP phone subnetwork, and traffic to and from this IP subnetwork is subject to LLQ.

The Cisco 7200 router routes between the primary and secondary IP subnetworks. The Mutilayer Switch Feature Card (MSFC) in the Catalyst 6500 switch is not currently used.

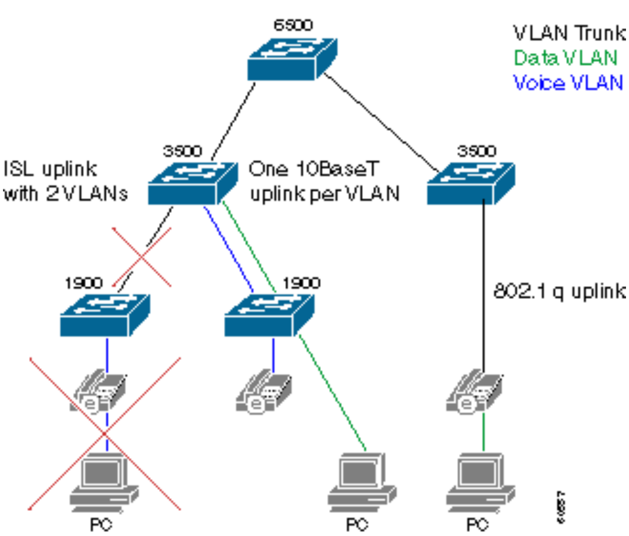
The Catalyst 3500XL and Catalyst 6500 switches have QoS features, but they are not currently enabled.

QoS in the Campus

The current campus design does not comply with the Cisco−recommended design guidelines for IP Telephony. These are some concerns about QoS:

- The broadcast domain is very large. Excessive broadcasts can affect the performance of IP phones, which have to process them.
- The Catalyst 1900 switches are not QoS−capable. If an IP phone and PC are connected to the same switch port, voice packets can be dropped if the PC receives data at a high rate.

Redesign parts of the campus infrastructure to achieve significant improvements. A hardware upgrade is not required. This diagram illustrates the principles behind the recommended redesign:



Phone and PC must be on separate ports on Catalyst 1900

The campus must be split into a voice VLAN and a data VLAN. Phones and PCs that connect to a Catalyst 1900 switch must now connect to different ports in order to achieve the VLAN separation. An additional uplink from each Catalyst 1900 switch to the Cisco 3500XL switch is added. One of the two uplinks is a member of the voice VLAN. The other uplink is a member of the data VLAN. Do not use InterSwitch Link (ISL) trunking as an alternative to two uplinks. This does not provide the voice and data traffic with separate queues. The GE links from the Catalyst 3500XL switch to the Catalyst 6000 switch must also be converted to 802.1q trunks so that both voice and data VLAN can be carried across this core switch.

Ports on the Catalyst 3500XL switch that are in the data VLAN have a default Class of Service (CoS) of zero. Ports that are members of the voice VLAN have a default CoS of 5. As a result, the voice traffic is correctly prioritized once it arrives at the Catalyst 3500 or Catalyst 6500 core. The Catalyst 3500 QoS switch port configurations vary slightly depending on which VLAN switch port is a member, as this example shows:

```
Interface fastethernet 0/1
description Port member of voice VLAN
switchport priority 5
switchport access vlan 1

Interface fastethernet 0/2
description Port member of data VLAN
switchport priority 0
switchport access vlan 2
```

You can connect a PC to the rear switch port on the IP phone in the rare case that IP phones connect directly to a Catalyst 3500XL switch. The IP phones connect to the switch by means of a 802.1q trunk in this case. This allows voice and data packets to travel on separate VLANs, and you can give packets the correct CoS at ingress. Replace Catalyst 1900 switches with Catalyst 3500XL switches or other QoS−capable switches as they reach end of life. This topology then becomes the standard method of connecting IP phones and PCs to the network. This scenario shows the Catalyst 3500XL switch QoS configuration:

```
Interface fastethernet 0/3
description Port connects to a 79xx IPhone
switchport trunk encapsulation dot1q
switchport priority extend 0
```

Finally, the two ports that connect to the two Cisco CallManagers should have the CoS hardcoded to 3. Cisco CallManager sets the IP precedence to 3 in all voice signaling packets. However, the link from the Cisco CallManager to the Catalyst 3500XL switch does not use 801.1p. Therefore, CoS value is forced at the switch as this example shows:

```
Interface fastethernet 0/1
description Port member of voice VLAN
switchport priority 3
switchport access vlan 1
```

The main hurdle with this design is that two switch ports are required at the Desktop. The Patrick campus might require an extra 400 switch ports for 400 IP phones. Additional Catalyst 3500XL switches must be deployed if sufficient ports are not available. Only one Catalyst 3500XL switch port is required for every two missing Catalyst 1900 switch ports.

The current ACU Catalyst 6500 switches have QoS capabilities, but they are not currently enabled. These modules are present in the ACU Catalyst 6000 switch with these queuing capabilities:

Slot	Module	Ports	RX Queues	TX Queues
1	WS−X6K−SUP1A−2GE	2	1p1q4t	1p2q2t
3	WS−X6408−GBIC	8	1q4t	2q2t
4	WS−X6408−GBIC	8	1q4t	2q2t
5	WS−X6248−RJ−45	48	1q4t	2q2t
15	WS−F6K−MSFC	0		

Complete these steps to activate the appropriate QoS features on the Catalyst 6000 switch:

1. Tell the switch to provide QoS on a per−VLAN basis with this command:

   ```
   Cat6K>(enable)set port qos 1/1−2,3/1−8,4/1−8 vlan−based
   ```
2. Tell the switch to trust the CoS values received from the Catalyst 3500XL switch with this command:

   ```
   Cat6K>(enable)set port qos 1/1−2,3/1−8,4/1−8 trust trust−cos
   ```

The CoS must now be set to differentiated services code point (DSCP) mapping. This is required because the Catalyst 6000 switch rewrites the DSCP value in the IP header based on the received CoS value. VoIP signaling packets must have a CoS of 3, rewritten with a DSCP of AF31 (26). RTP packets must have a CoS of 5, rewritten with a DSCP of EF (46). Issue this command:

```
Cat6K>(enable)set qos cos−dscp−map 0 8 16 26 32 46 48 56
```

Use this example to verify the CoS−to−DSCP mapping.

```
Cat6K> (enable) show qos map run COs−DSCP−map
CoS − DSCP map:
CoS DSCP
−−− −−−−
 0   0
 1   8
 2   16
 3   26
 4   32
 5   46
 6   48
 7   56
```

Configure the MSFC to route between the various IP subnetworks.

QoS in the RNO

The current RNO design does not comply with Cisco−recommended design guidelines for IP Telephony. These concerns exist in regards to QoS:

- LLQ is not applied on the Cisco ACU 7200 series WAN router.
- The Patrick and Aquinas campuses connect to the RNO by means of ATM switched VCs (SVCs). LLQ is not supported on SVCs.

A Fast Ethernet−attached Cisco 7200 router connects the campus to an RNO by means of a 34 Mbps E4 ATM link. Traffic can potentially queue up outbound on the 34M links because of the 4M versus 100M speed mismatch. Therefore, it is necessary to prioritize the voice traffic. Use LLQ. The Cisco 7200 router configuration is similar to this example:

```
class−map VoiceRTP
match access−group name IP−RTP

policy−map RTPvoice
class VoiceRTP
priority 10000

interface ATM1/0.1 point−to−point
description ATM PVC to RNO
pvc 0/100
tx−ring−limit 3
service−policy output RTPvoice

ip access−list extended IP−RTP
```

```
deny ip any any fragments
permit udp any range any range 16384 32768 precedence critical
```

The bandwidth allocated to LLQ must be *N x 24Kbps*, where N is the number of simultaneous G.729 calls.

Set up one PVC from each of the Patrick and Aquinas Cisco 7200 routers to the AARNet router. ATM SVCs in the Victoria RNO do not support LLQ, as it is based on Classical IP over ATM (RFC 1577). The other universities in the Victoria RNO can continue to use RFC 1577 for now. However, eventually replace the Classical IP over ATM infrastructure.

Gateways

Each of the ACU campuses has a Cisco 3640 router that acts as an H.323 gateway. These gateways connect to the PSTN by means of ISDN. The number of Primary Rate Interfaces (PRIs) and B−channels depends on the size of the campus. This table lists the number of PRIs and B−channels for each campus:

Campus	PRI Quantity	B−channel Quantity
Mount Saint Mary	2	30
MacKillop	2	50
Patrick	2	50
Aquinas	1	20
Signadou	1	20
McAuley	1	30

These gateways are used only as secondary gateways for DOD (Direct Outward Dialing). The AARNet gateways are the primary gateways. The ACU gateways are always used for DID (Direct Inward Dialing).

Dial Plan

The dial plan is based on 4−digit extension numbers. The extension is also the last four digits of the DID number. This table lists the extension ranges and DID numbers for each campus:

Campus	Extension	DID
Mount Saint Mary	9xxx	02 9764 9xxx
MacKillop	8xxx	02 9463 8xxx
Patrick	3xxx	03 8413 3xxx
Aquinas	5xxx	03 5330 5xxx
Signadou	2xxx	02 6123 2xxx
McAuley	7xxx	07 3354 7xxx

A simple `num−exp` entry on the gateways truncate the DID number to the 4−digit extension before it passes it on to Cisco CallManager. For example, the Patrick campus gateway has this entry:

```
num−exp 84133... 3...
```

Users dial zero to select an outside line. This leading zero is passed on to the gateway. A single POTS dial peer routes the call out the ISDN port based on the leading zero.

```
Dial-peer voice 100 pots
destination-pattern 0
direct-inward-dial
port 2/0:15
```

Incoming calls use this num−exp entry to transform the called party number to a 4−digit extension. The call then matches both VoIP dial peers. Based on the lower preference, it prefers this route to the Cisco CallManager subscriber:

```
dial-peer voice 200 voip
preference 1
destination-pattern 3...
session target ipv4:172.168.0.4

dial-peer voice 201 voip
preference 2
destination-pattern 3...
session target ipv4:172.168.0.5
```

Cisco CallManager

Each of the campuses has a cluster that consists of two Cisco CallManager servers. The Cisco CallManager servers are a mix of Media Convergence Server 7835 (MCS−7835) and Media Convergence Server 7820 (MCS−7820). Both servers ran version 3.0(10) at the time of this publication. One Cisco CallManager is the *publisher* and the other Cisco CallManager is the *subscriber*. The subscriber acts as the primary Cisco CallManager for all IP phones. This table lists the hardware deployed at each campus:

Campus	Platform	CallManagers
Mount Saint Mary	MCS−7835	2
MacKillop	MCS−7835	2
Patrick	MCS−7835	2
Aquinas	MCS−7820	2
Signadou	MCS−7820	2
McAuley	MCS−7835	2

Each cluster is configured with two regions:

- One for intracampus calls (G.711)
- One for intercampus calls (G.729)

Location−based CAC is not appropriate for ACU because all IP phones served by each cluster are on a single campus. There are merits to a gatekeeper−based CAC for intercampus calls, but this is not currently implemented. However, there are plans to do so in the near future.

Each Cisco CallManager is configured with 22 H.323 gateways. This is composed of intercluster trunks to the five other Cisco CallManager clusters, six AARNet PSTN gateways, and one ACU gateway at each campus.

H.323 Device Type	Quantity
Intercampus CallManager	2 x 5 = 10
AARNet PSTN Gateway	6
ACU PSTN Gateway	6

Total:	*22*

Route lists and route groups are used to rank the PSTN gateways. For example, this table shows how calls from the Patrick Cisco CallManager in Melbourne to the Sydney PSTN can use the four gateways to tie the calls together with a route group.

Gateway	Priority
AARNet Sydney	1
ACU Sydney	2
AARNet Melbourne	3
ACU Melbourne	4

The Cisco CallManagers are configured with approximately 30 route patterns, as this table shows. The route patterns are designed so there are specific matches for all domestic Australian numbers. This way, the users do not have to wait for the interdigit timeout to expire before Cisco CallManager initiates the call. The wildcard character "!" is used only in the route pattern for international numbers. Users must wait until the interdigit timeout (default 10 seconds) expires before the call progresses when they dial an international destination. Users can also add the route pattern "0.0011!#". Users can then enter a "#" after the last digit to indicate to Cisco CallManager that the dialed number is complete. This action expedites international dialing.

Route Pattern	Description
0.[2−9]XXXXXXX	Local call
0.00	Emergency call − if user forgets to dial 0 for outside line
0.000	Emergency call
0.013	Directory assistance
0.1223	
0.0011!	International calls
0.02XXXXXXXX	Calls to New South Wales
0.03XXXXXXXX	Calls to Victoria
0.04XXXXXXXX	Calls to cell phones
0.07XXXXXXXX	Calls to Queensland
0.086XXXXXXX	Calls to Western Australia
0.08XXXXXXXX	Calls to South Australia and Northern Territory
0.1[8−9]XXXXXXXX	Calls to 1800 xxx xxx and 1900 xxx xxx
0.1144X	Emergency
0.119[4−6]	Time and Weather
0.1245X	Directory
0.13[1−9]XXX	Calls to 13xxxx numbers
0.130XXXXXXX	Calls to 1300 xxx xxx numbers

2[0−1]XX	Intercluster calls to Signadou
3[0−4]XX	Intercluster calls to Patrick
5[3−4]XX	Intercluster calls to Aquinas
7[2−5]XX	Intercluster calls to McAuley
8[0−3]XX	Intercluster calls to MacKillop
9[3−4]XX	Intercluster calls to Mount Saint Mary
9[6−7]XX	Intercluster calls to Mount Saint Mary

The number of gateways, route groups, route lists, and route patterns configured on the ACU Cisco CallManagers has the potential to grow to a large number. If a new RNO gateway is deployed, all five Cisco CallManager clusters must be reconfigured with an additional gateway. Even worse, hundreds of gateways need to be added if ACU Cisco CallManagers route VoIP calls directly to all other universities and bypass the PSTN altogether. Clearly this does not scale very well.

The solution is to make the Cisco CallManagers gatekeeper−controlled. You must only update the gatekeeper when a new gateway or Cisco CallManager is added somewhere in the AARNet. Each Cisco CallManager must have only the local campus gateway and the anonymous device configured when this happens. You can think of this device as a point−to−multipoint trunk. It removes the necessity for the meshed PPP trunks in the Cisco CallManager dial plan model. A single route group points to the anonymous device as the preferred gateway and to the local gateway as the backup gateway. The local PSTN gateway is used for certain local calls and also for general off−net calls if the gatekeeper becomes unavailable. Currently, the anonymous device can be either intercluster or H.225, but not both at the same time.

Cisco CallManager needs fewer route patterns with a gatekeeper than it has now. In principle, the Cisco CallManager needs only a single route pattern of "!" pointing to the gatekeeper. In reality, the manner in which calls are routed needs to be more specific for these reasons:

- Some calls (such as calls to 1−800 or emergency numbers) need to be routed through a geographically local gateway. Someone in Melbourne who dials the police or a restaurant chain such as Pizza Hut does not want to be connected to the police or the Pizza Hut in Perth. The specific route patterns are needed that point directly to the local campus PSTN gateway for these numbers.

  Universities that plan to perform future IP Telephony deployments can choose to rely solely on the AARNet gateways and not administer their own local gateways. These numbers must have a virtual area code prepended by Cisco CallManager before sending it to the gatekeeper in order to make this design work for calls that need to be dropped off locally. For example, Cisco CallManager can prepend 003 to calls from a Melbourne−based phone to the Pizza Hut 1−800 number. This allows the gatekeeper to route the call to a Melbourne−based AARNet gateway. The gateway strips off the leading 003 before it places the call into the PSTN.
- Use route patterns with specific matches for all domestic numbers in order to avoid having the user wait for the interdigit timeout before the call is initiated.

This table shows the route patterns for a gatekeeper−controlled Cisco CallManager:

Route Pattern	Description	Route	Gatekeeper
0.[2−9]XXXXXXX	Local call	Route list	AARNet
0.00	Emergency call	Local gateway	None

0.000	Emergency call	Local gateway	None
0.013	Directory assistance	Local gateway	None
0.1223		Local gateway	None
0.0011!	International calls	Route list	AARNet
0.0011!#	International calls	Route list	AARNet
0.0[2−4]XXXXXXXX	Calls to New South Wales, Victoria, and cell phones	Route list	AARNet
0.0[7−8]XXXXXXXX	Calls to South Australia, Western Australia, and Northern Territory	Route list	AARNet
0.1[8−9]XXXXXXXX	Calls to 1800 xxx xxx and 1900 xxx xxx	Local gateway	None
0.1144X	Emergency	Local gateway	None
0.119[4−6]	Time and weather	Local gateway	None
0.13[1−9]XXX	Calls to 13xxxx numbers	Local gateway	None
0.130XXXXXXX	Calls to 1300 xxx xxx numbers	Local gateway	None
[2−3]XXX	Calls to Signadou	Route list	ACU
5XXX	Calls to Aquinas	Route list	ACU
[7−9]XXX	Calls to McAuley, MacKillop, and Mount Saint Mary	Route list	ACU

The gatekeeper routes international calls, which are not sent through the local gateway. This is significant because AARNet can deploy international gateways in the future. If a gateway is deployed in the United States, a simple gatekeeper configuration change allows universities to place calls to the US at US domestic rates.

The gatekeeper performs intercluster call routing based on the 4−digit ACU extension. This address space most likely overlaps with other universities. This dictates that ACU administer its own gatekeeper and use the AARNet gatekeeper as a *directory gatekeeper*. The gatekeeper column in this table indicates whether the call routing is performed by the ACU gatekeeper or the AARNet gatekeeper.

Note: The sole caveat with the proposed gatekeeper solution is that the anonymous device can currently be either intercluster or H.225, but not both at the same time. Cisco CallManager relies on the gatekeeper to route calls to both gateways (H.225) and other Cisco CallManagers (intercluster) with the proposed design. The workaround for this issue is to either not use the gatekeeper for intercluster routing or to treat all calls via the gatekeeper as H.225. The latter workaround means that some supplementary features might be unavailable on intercluster calls.

Voice Mail

ACU had three Active Voice Repartee OS/2−based voice mail servers with Dialogic phone boards prior to the migration to IP Telephony. The plan is to reuse these servers in the IP Telephony environment. When implemented, each Repartee server connects to a Cisco CallManager by means of a simplified message desk interface (SMDI) and a Catalyst 6000 24−Port Foreign Exchange Station (FXS) card. This provides voice mail for three of the six campuses, which leaves three campuses without voice mail. It is not possible to properly share one Repartee server between users on two Cisco CallManager clusters because there is no way to propagate the message waiting indicator (MWI) across the intercluster H.323 trunk.

ACU might purchase three Cisco Unity servers for the campuses that remain. These servers are Skinny−based, so no gateways are required. This table lists the voice mail solutions in the event that ACU purchases the additional voice mail servers:

Campus	Voice Mail System	Gateway
Mount Saint Mary	Active Voice Repartee	Catalyst 6000 24−port FXS
MacKillop	Active Voice Repartee	Catalyst 6000 24−port FXS
Patrick	Active Voice Repartee	Catalyst 6000 24−port FXS
Aquinas	Cisco Unity	
Signadou	Cisco Unity	
McAuley	Cisco Unity	

The six voice mail servers operate as isolated voice mail islands in this plan. There is no voice mail networking.

Media Resources

Hardware digital signal processors (DSPs) are not currently deployed at ACU. Conferencing uses the software−based Conference Bridge on the Cisco CallManager. Intercluster conferencing is not currently supported.

Transcoding is currently not required. Only G.711 and G.729 coder−decoders are used, and they are supported by all deployed end devices.

Fax and Modem Support

Fax and modem traffic is not currently supported by the ACU IP Telephony network. The university plans to utilize the Catalyst 6000 24−Port FXS card for this purpose.

Software Versions

This table lists the software versions ACU used at the time of this publication:

Platform	Function	Software Version
CallManager	IP−PBX	3.0(10)
Catalyst 3500XL	Distribution switch	12.0(5.1)XP
Catalyst 6500	Core switch	5.5(5)
Catalyst 1900	Wiring closet switch	
Cisco 7200 processor	WAN router	12.1(4)
Cisco 3640 router	H.323 gateway	12.1(3a)XI6

Related Information

- **Voice Technology Support**
- **Voice and IP Communications Product Support**
- **Recommended Reading: Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation − Cisco Systems**

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