JANET Voice Strategy

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Final Version

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CONTENTS

Executive Summary

This document describes UKERNA's strategy for the provision of voice services on JANET, as informed by a requirements analysis. The strategy can be summarised in the objectives listed below in section 1.1.

1. Introduction

In response to growing interest from the JANET community in using JANET to carry voice traffic, UKERNA formed a JANET Voice Advisory Group early in 2005. The group is charged with helping UKERNA to understand the application services support requirements on JANET. A requirements analysis was initiated in August 2005, and as a result a strategy for the support of voice on JANET has been produced which is presented in this document.

Many JANET connected organisations operate VoIP (Voice over IP) systems; some now use IP-based internal telephony systems, rather than traditional analogue private exchanges. One potential use of VoIP would be to provide inter-organisation VoIP connectivity across JANET, as a direct replacement for regular telephony links. It may also be feasible to provide centrally operated services, such as VoIP-based audio conference bridges, presence services and instant messaging.

One area of interest, particularly to smaller organisations, is the possibility of providing centrally purchased bulk telephony deals, which would allow an organisation to route all its telephony calls via JANET. Those calls to other participating VoIP sites would route directly over JANET; others would route to a gateway to the PSTN (Public Switched Telephony Network).

1.1 Strategic Objectives

Throughout this document each area pertains to a specific objective for the provision of voice services on JANET. These objectives are summarised below:

- **UKERNA will deploy services to promote JANET as a means to carry JANET community voice traffic.**
- **UKERNA will build and deploy a trial infrastructure based on a central voice switch with downloadable software voice clients. If successful, UKERNA will develop a service based on this technology. (Section 2.1.1)**
- **UKERNA will deploy a JANET ENUM server(s), which will form the ENUM tree e164.ja.net. Based upon usage and community feedback, UKERNA may develop this technology as a service. (Section 2.1)**
- **Through the JANET Voice Advisory Group, and community feedback, UKERNA will assess and where relevant provide resources to support any additional service requirements. (Section 2.2)**
- **UKERNA will work collaboratively through both TERENA and Internet2. Where possible, VoIP equipment will interoperate throughout the international community. (Section 2.2.4)**
- **UKERNA will investigate the potential of providing VoIP services using the JANET Roaming service. Where relevant, JANET voice services will utilise the JANET Roaming service. (Section 2.2.5)**

- **UKERNA will undertake investigation work with regulatory bodies and telephony providers to understand how PSTN and/or telephony peering services may be provided to the JANET community. Where relevant, UKERNA will provide or make arrangements for both PSTN and/or telephony peering facilities for the JANET community. (Section 2.3)**
- **Through community, NREN (National Research and Education Network) and regulatory consultation UKERNA will provide or recommend a numbering scheme for any voice services. (Section 2.5)**
- **UKERNA will produce documentation aimed at providing best practice, recommendations and factsheets for the JANET community on the use of voice. (Section 2.6)**

1.2 Consultation Process

To understand the precise requirements of JANET users for voice services over JANET, UKERNA undertook a requirements gathering exercise. The key elements of this exercise comprised:

- a requirements survey issued to all JANET connected organisations
- an invitation to comment on any aspect of the draft JANET Voice Strategy.

This first version of the voice strategy has been informed by the requirements analysis.

2. THE JANET VOICE STRATEGY

2.1 Service Model

A key aspect of the voice strategy has been to decide whether any JANET voice support services are provided on an end user basis or based around the standard JANET model, where central services are provided to connected organisations, and those organisations then provide the service to their end users as they see fit.

With an estimated 16 million users that have access to JANET, providing a service to end users directly is likely to be costly in both financial and effort terms. In addition, it may complicate the way individual JANET connected organisations implement their network access and security policies.

However, where an organisation has no installed VoIP services, direct access for end users could be advantageous, and may in fact be necessary for some possible complementary services such as instant messaging or presence, where a user indicates their physical or virtual location and communication preference using an online register.

It is clear that there are two emerging user models: the provision of services directly to the user, and the provision of services aimed at increasing the connectivity and reach of voice services which are provided and supported by individual organisations. In the former scenario, users would be quite conscious of the fact that they were using a VoIP service while in the latter they may well be ignorant of the fact: for example, the user would make the call, perhaps from the same phone they have used for many years, and the telephony equipment would make the decision to route the call over VoIP in much the same way as it might choose to route a call via an alternative carrier. We have named these two user models the *active model* and the *passive model*. These models are described in detail in the following sections.

2.1.1 Active Model

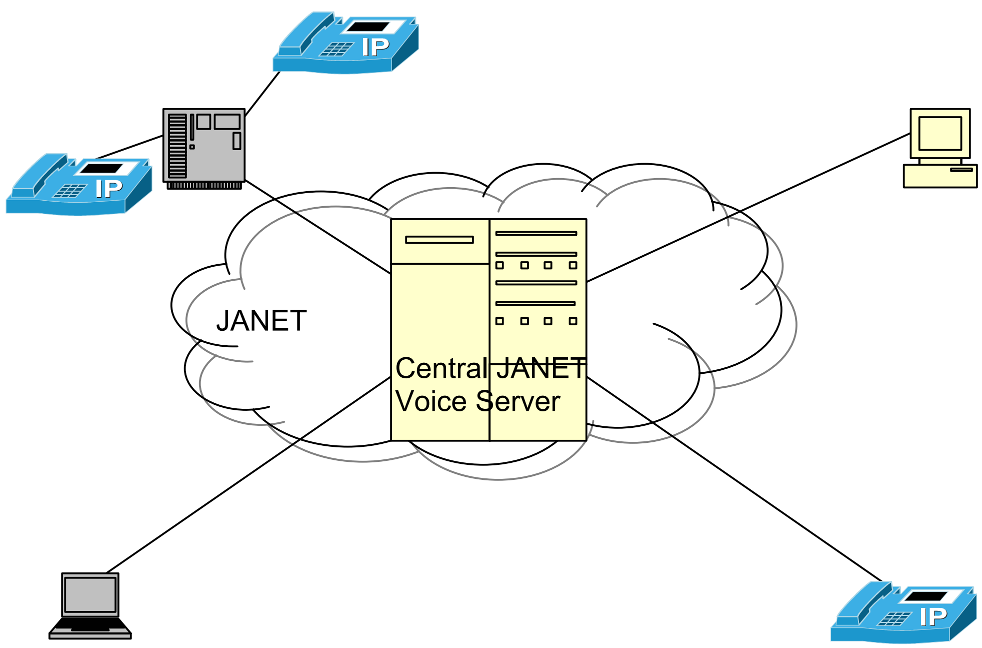
In the active model, equipment owned and managed centrally is installed in the JANET backbone, and users or organisations would register with it when they are capable of accepting VoIP calls. For users, this might be when their VoIP equipped computer starts up; for an organisation this may happen when its telephony system starts up.

The latter may remain active for extended periods (for example, until the system reboots, or there is a loss of network connectivity). A user registration would most likely be for a single voice endpoint (a single telephone number or other VoIP address); an organisation's registration would cover the range of telephone numbers that the system is capable of handling.

This central system would then take an active part in establishing calls, and may participate in the audio stream between the two voice endpoints. For example, a JANET voice server, such as a SIP-based telephony system, could be established with which both end user and organisational VoIP systems could register, indicating their ability to receive VoIP calls.

When making a call, the endpoint would make a call setup request to the central server, which would attempt to match the call against its call routing table, and connect it accordingly. This might be by connecting the call through the server itself, or returning sufficient information for the two endpoints to connect the call directly.

If no match could be found, the server would indicate this back to the endpoint, and then local policy would dictate how the call was further handled (for example, default to placing the call over the PSTN).



Strategic Objective:	UKERNA will build and deploy a trial infrastructure based on a central voice switch with downloadable software voice clients. If successful UKERNA will develop a service based on this technology.

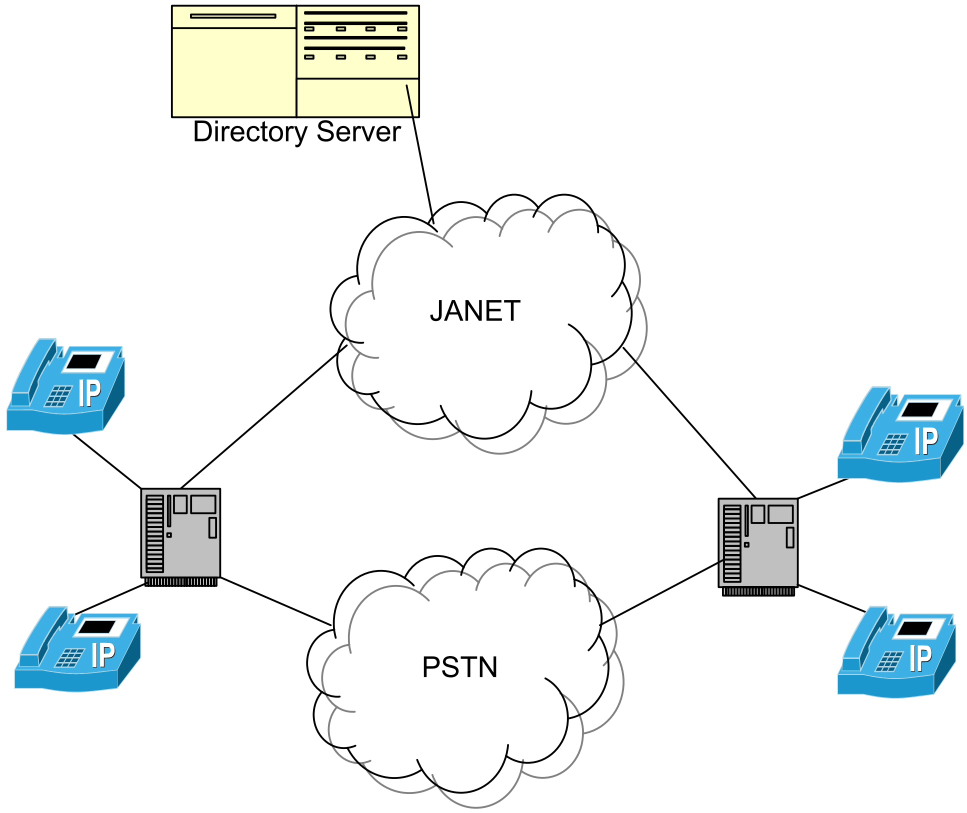
2.1.2 Passive Model

In the passive model, instead of providing active equipment, voice interconnectivity is provided simply by facilitating the routing of call setup requests and having no central equipment brokering or connecting the call. This model is essentially about providing a telephone directory for the voice systems at JANET connected organisations to consult. Specifically, this would not be a directory designed to be consulted by humans.

For example, a private JANET directory could be established, mapping the standard UK national telephone numbers of JANET connected organisations to the IP address of their VoIP system. Before dialling the number over the PSTN, a VoIP enabled telephone system would first check the JANET directory, and if it found a match (either exact or perhaps a DDI (direct dial-in) prefix) then it would route that call over IP.

The IETF (Internet Engineering Task Force) ENUM standard (described in RFC2916) is an example of a passive architecture. Information about the reachability of standard E.164 telephone numbers is stored in the DNS (Domain Name System), and can be queried in the same fashion as any other DNS query. A suitably equipped VoIP system consults the DNS and, if VoIP reachability records are returned, the information in these records can be used to route a call over IP.

The call routes directly from the initiating system to the system indicated in the DNS – it does not pass through intermediate equipment at the application (VoIP) layer.



Strategic Objective:	UKERNA will deploy a JANET ENUM server(s), which will form the ENUM tree e164.ja.net. Based upon usage and community feedback, UKERNA may develop this technology as a service.

2.2 Additional Services

In addition to providing basic VoIP connectivity, there is a requirement to provide other complementary services. Such services would almost certainly require active equipment to be provided, which would not necessarily have to be tied to any equipment provided as part of the provisioned VoIP model.

Strategic Objective:	Through the JANET Voice Advisory Group and community feedback, UKERNA will assess and where relevant provide resources to support any additional service requirements.

2.2.1 Audio Conferencing

Many organisations make use of multi-party conference calls. Within the same organisation, this is usually possible using the local telephony system; often, however, there are restrictions on the number of external parties that can be connected. As a result, use is often made of third party 'conference bridges', the use of which attracts a fee.

As part of ongoing technology investigation work, UKERNA will trial a JANET conference bridge. It is likely that this will span both PSTN and VoIP users.

2.2.2 JANET Videoconferencing Integration

As an extension of the audio conferencing scenario, it may be useful to provide a VoIP gateway into the JANET Videoconferencing Service, so that parties can join a videoconference via audio if there are no suitable videoconferencing facilities available. The JANET community has a wide deployment of H.323 videoconferencing endpoints, the majority of which are able to make voice calls. In some instances it may be useful for these to be used as VoIP endpoints, in particular those endpoints used in a desktop environment.

Where possible UKERNA will ensure that voice services are interoperable with existing JVCS infrastructure.

2.2.3 Presence/Instant Messaging

The popularity of instant messaging and presence services, such as ICQ, MSN Messenger and AOL Instant Messenger, has grown steadily in recent years, and many organisations now use instant messaging systems as part of their business communications. Providing a dedicated service for JANET may be a logical extension of a VoIP solution where service is provided to end users, as many vendors include this functionality in their end user solutions.

UKERNA will undertake development work to further understand any requirements for presence and instant messaging. Based upon this work, UKERNA will include these services as part of an overall voice service.

2.2.4 Inter-NREN VoIP

Extending the service model to include co-operation with other NRENs to establish VoIP connectivity on a formal, managed basis has the potential to provide an area of cost saving for JANET connected organisations. JANET already participates in videoconferencing interconnection with other NRENs, so it may be that this could be achieved with relative ease.

Strategic Objective:	UKERNA will work collaboratively through both TERENA and Internet2. Where possible VoIP equipment will interoperate throughout the international community.

2.2.5 ‘Roaming’ VoIP

As VoIP simply relies on an IP connection, it is possible for users to connect to their home voice system from remote locations where IP network connectivity is available. Depending upon the chosen configuration, users could be reachable on the same phone number as they use when in the office. They could contact colleagues simply by using their extension number, access voicemail and make external calls via the voice system at their home site.

Whilst being convenient, this clearly has security implications. Therefore care needs to be taken when formulating an architecture to make this possible – whether at a site level, or at a JANET level if an end-user based active model were to be implemented.

Solutions at the site level might require the establishment of a VPN connection to site before access to the voice system is possible, or running a second voice system for external users to connect to. This system might well have fewer capabilities (e.g. no external calls).

JANET is currently developing a JANET Roaming service based on the success of the now completed LIN (Location Independent Networking) project trial service. It may therefore be feasible to extend the Roaming service to support VoIP 'Roaming'.

Strategic Objective:	UKERNA will investigate the potential of providing VoIP services using the JANET Roaming service. Where relevant, JANET voice services will utilise the JANET Roaming service.

2.3 PSTN Gateway

A logical extension of a VoIP service interconnecting JANET connected organisations is to provide the ability to make calls beyond the VoIP service to regular telephones on the PSTN. This would in effect allow a JANET connected organisation to use JANET for all its voice connectivity.

Whilst at first sight this might seem attractive, the regulatory requirements for providing such a service are onerous (see Section 3 below). In addition, the billing function that would be required for UKERNA to recover call charges from organisations would need a substantial amount of staff and resources to operate.

However, even though it is unlikely that UKERNA would be in a position to operate a PSTN gateway directly, it may be possible to outsource this service element to a third party telecommunications provider.

Strategic Objective:	UKERNA will undertake investigation work with regulatory bodies and telephony providers to understand how PSTN and/or telephony peering services may be provided to the JANET community. Where relevant, UKERNA will provide or make arrangements for both PSTN and/or telephony peering facilities for the JANET community.

2.4 Interconnection with Other Voice Networks

While providing a full PSTN service may be beyond the scope of VoIP on JANET, it may be possible to establish private interconnections with other voice telephony providers, particularly if this were possible using VoIP rather than standard telephony circuits.

For example, it may be possible to engineer a link to a mobile or telephony network provider, for the sole purpose of exchanging traffic between JANET VoIP users and customers of that mobile network. As this is not an arrangement to provide full telephony service, the regulatory requirements are not as onerous.

This sort of interconnection would, of course, be subject to a commercial decision on the part of the other network operator. The possibility of this kind of interconnection requiring substantial amounts of resource for billing may also make the service impractical.

UKERNA will investigate interconnecting with telephony providers.

2.5 Numbering and Addressing

The numbering and addressing of any service is an important factor. Organisations will already have telephone numbers from the UK numbering space which conform to the European E.164 standard. JANET and other NRENs operate a private E.164 scheme for the videoconferencing service; users of Internet based voice or instant messaging services might be known by numbers from further private numbering schemes, or by nicknames or e-mail addresses or e-mail address-like identifiers.

It would be possible to operate a JANET VoIP service using any, all or a combination of these methods, perhaps including a degree of aliasing (e.g. relating an e-mail address to an E.164 number), or even a totally different scheme.

The chosen solution has to be as straightforward as possible, and capable of scaling to many millions of users.

Strategic Objective:	Through community, NREN and regulatory consultation UKERNA will provide or recommend a numbering scheme for any voice services.

2.6 Provision of advice

UKERNA plans to provide comprehensive advice on a range of topics associated with VoIP networks. This advice will be provided in the form of factsheets and case studies, for which the JANET web server, www.ja.net, is expected to be the principle outlet.

The initial set of topics that have been identified are:

- Glossary of VoIP related terms
- VoIP: What is it and why?
- Voice and VoIP Regulatory Issues
- Addressing, numbering and directories
- VoIP deployments within the campus
- Interconnecting traditional telephony systems and VoIP
- Interconnecting IP telephony systems
- VoIP on network operating systems
- The SKYPE VoIP system.

Strategic Objective:	UKERNA will produce documentation aimed at providing best practice, recommendations and factsheets for the JANET community on the use of voice.

2.7 IP Network Issues

Clearly the standard of any VoIP service will be directly related to the reliability of the underlying IP network. Once calls are routed over VoIP outside the campus, the standard is no longer based on a single network – particularly, the network is no longer under the direct control of the local organisation.

The intermediate networks, and the network of the organisation a VoIP call is destined for, must therefore be capable of carrying VoIP traffic appropriately. This has implications in the following areas:

2.7.1 Availability

Users seldom, if ever, pick up a standard telephone and do not receive a dialling tone that indicates the ability to attempt to make a call. More often (although still very infrequently) calls do not connect properly, and have to be re-tried – sometimes immediately, sometimes a little later – before they connect successfully.

It is sensible to assume, therefore, that for a VoIP service to be an integral part of an organisation's telecommunications strategy, the end to end network path should be available to the same degree. (Or if not, for capability to be built into the model to fall back to the PSTN where possible.)

2.7.2 Performance

Once a call is established, voice traffic is particularly sensitive to IP packet loss, out of sequence packet ordering and varying delays in the arrival of packets at the destination (known as 'jitter').

Experience tends to show that VoIP can operate successfully over IP networks as they are today, operating a 'best efforts' service, where no guarantee is made about how a packet will be treated en-route to its destination, or even that it will get there at all. For example, many remote workers maintain successful VoIP connectivity to their offices via ADSL services over the public Internet.

This level of operation may well be inappropriate for a centrally supported and coordinated voice service on JANET.

JANET is currently entering the second phase of an IP QoS (quality of service) pilot. During phase one it was shown that voice and video traffic can benefit from IP QoS by receiving priority treatment within the network. This makes it more likely that packets will be delivered, and delivered promptly with a lower rate of jitter (although the technologies investigated so far do not absolutely guarantee delivery of a data packet – other factors such as the congestion level on a network may still cause problems).

Work from the IP QoS project is likely be useful in the support of any voice service deployed on JANET.

2.7.3 Security

As with any network application, security is a major concern for VoIP deployments, particularly as there is potential for financial loss (such as unauthorised calls over an organisation's PSTN lines).

Securing telephony equipment from this sort of intrusion is not a new concept in the voice environment; however the connection of a telephony system to IP networks provides a new source for this sort of intrusion attempt. Authorisation and authentication have a key role in any VoIP strategy.

In addition, many VoIP systems are based on standard server platforms, in much the same way as any other network service such as a web or e-mail server. These sorts of systems are therefore open to exactly the same cracking attempts as any other network server and so require the same level of securing, updating and monitoring at the IP level, in addition to traditional voice security methods.

> **UKERNA will ensure that any voice services are engineered to utilise the network in an efficient, reliable and secure manner.**

3. REGULATORY CONSIDERATIONS

Regulations on the types of voice service that can be offered are clearly defined between a Publicly Available Telephony Service (PATS) and an Electronic Communications Service (ECS). During 2005 these regulations have been revised by OFCOM and clearly state that if any of the following services are provided by a voice service provider, the service becomes a PATS:

- a service available to the public
- a service for originating and receiving national and international phone calls
- a service that provides access to the emergency services
- a service available through a number or numbers in a national or international telephone numbering plan.

The provision of a PATS brings with it a number of strict criteria:

- proper and effective functioning of the network
- emergency call numbers
- emergency planning
- operator assistance, directories and Directory Enquiry Facilities
- transparency and publication of information
- metering and billing
- itemised bills
- non-payment of bills
- measures for end-users with disabilities

As a provider of a best effort IP network (JANET) it is unlikely in the current environment that UKERNA would be in a position to provide such services.

UKERNA is working with OFCOM where relevant to ensure that any voice regulations are adhered to.

4. FURTHER INFORMATION

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