BGAN IP Voice with the BGAN Radio Module

Application Note

Version 1.2

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Contents

1: Version History

Version	Date	Description
1.2	21-Mar-2018	Changed name of the document to 'BGAN IP Voice with the BGAN Radio Module - Application Note', to take account of it focusing on integration with the BGAN Radio Module (BRM).

2: Introduction

Note: The content of this document is Inmarsat proprietary and confidential, and as such external distribution is available only to Inmarsat-approved Value Added Manufacturer Partners involved in development using the BGAN Radio Module.

The **BGAN Radio Module** (**BRM**) does not offer any Circuit Switched (CS) Services. In order to allow users to make voice calls, Inmarsat have decided to implement a Voice-over-Internet Protocol (VoIP) service specifically for terminals based on BRM, called **BGAN IP Voice**. This concept is therefore similar to the Voice-over-LTE (VoLTE) offering on terrestrial 4G networks. Initially, BGAN IP Voice shall offer a single voice call per BRM-based terminal.

Note: The BGAN IP Voice service has not yet been finalised, so this document may be subject to further updates.

3: Service Overview

BGAN IP Voice provides a single voice channel as a Voice over Internet Protocol (VoIP) service, which is provided via Inmarsat's own VoIP service platform.

Terminals support provisioning of the following Voice services:

Note: BGAN IP Voice is currently only available to **Land-based** User Terminals.

> Capability to make incoming and outgoing calls

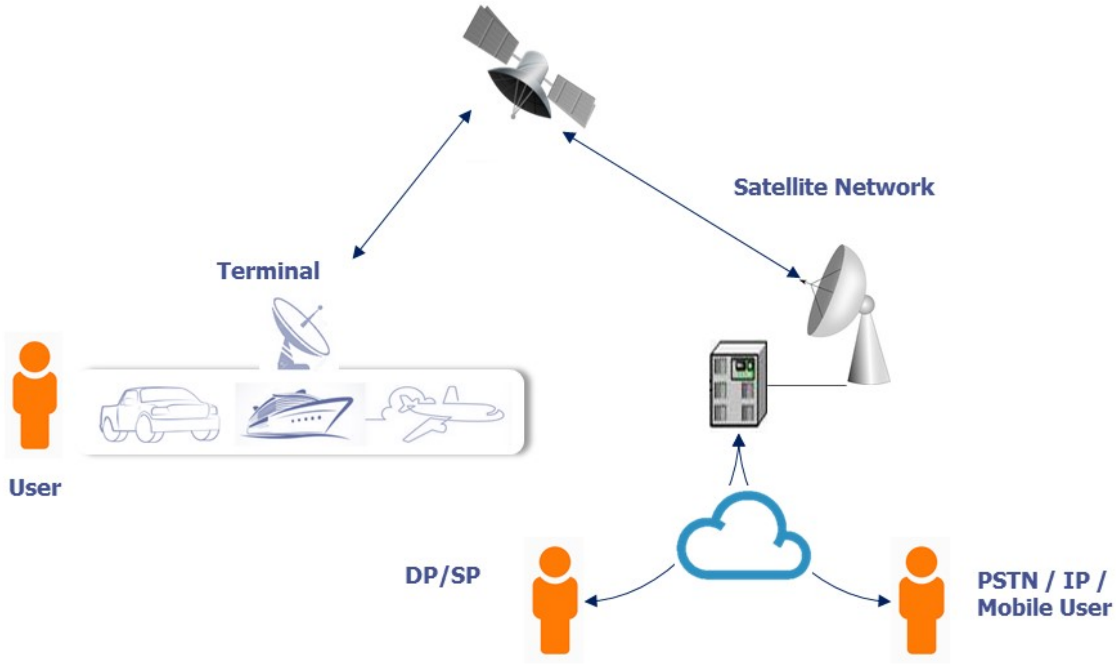
> Short-codes

> Post-pay

> Voicemail

> Supplementary Services

3 depicts the network architecture of the BGAN IP Voice service for Terminals.



Value Added Reseller (**VAR**): Provisioning partner (Service Provider (SP) or Distribution Partner (DP)) who provisions voice services for a SIM module on a BRM-based Terminal.

User: Represents the individual(s) that use(s) a BRM-based terminal for making and receiving voice calls and accessing other voice services.

Terminal/**User Terminal** (**UT**): Represents a BGAN product that has a BRM module inside as the core technology to provide the satellite modem capability.

Fixed Line and Mobile User: Fixed line and Mobile users are the calling or receiving party that the satellite service user is calling or receiving calls from.

Notes:

For BGAN IP Voice, Inmarsat offers the standard 99% availability Service Level Agreement (SLA).

VARs and VAMs will be notified in the usual fashion by Inmarsat's Global Customer Operations (GCO) if the service is ever unavailable.

VARs are advised to ensure appropriate alerts are provided to their BGAN IP Voice subscribers if the service is unavailable.

4: BGAN IP Voice Architecture

Inmarsat's BGAN IP Voice provides high quality voice service for users and is equally easy to provision for Value Added Resellers (VARs). Inmarsat's BGAN Core Network provides basic IMS (IP Multimedia Subsystem) functions for the hosted service.

The Core Network Nodes that support BGAN IP Voice are described in *Figure 1*.

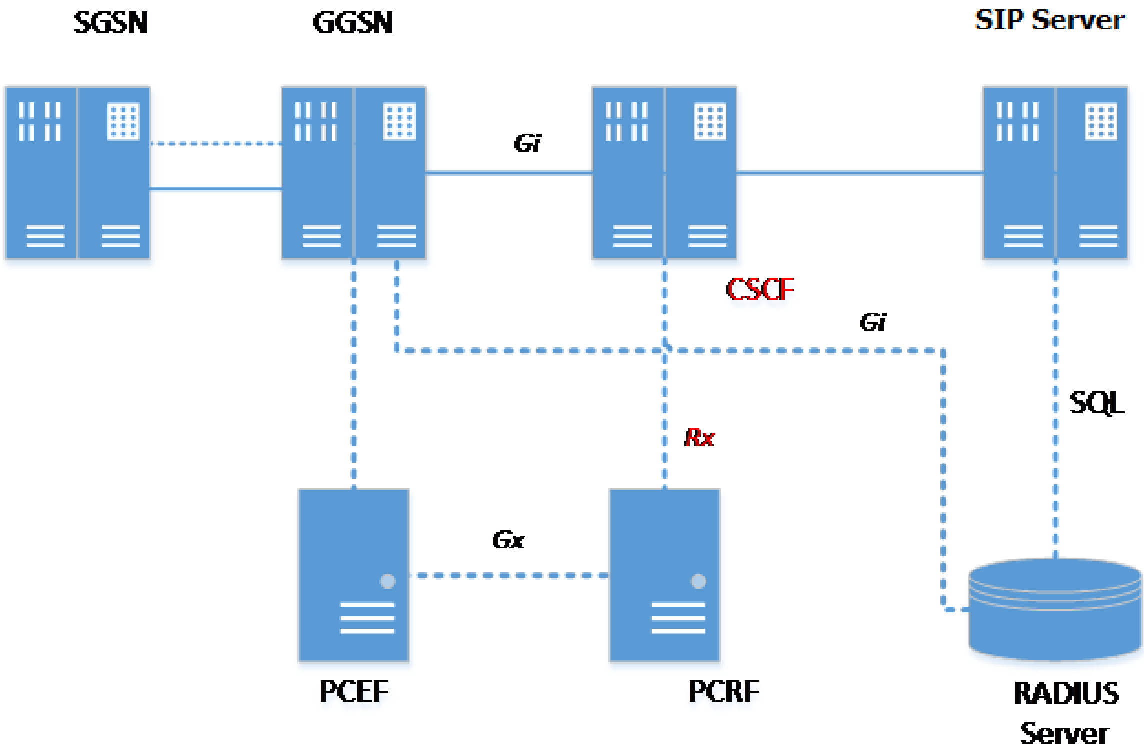


Figure 1. BGAN IP Voice Network Architecture

5: Implementation Guidelines

For test purposes, VAMs can use **Asterisk** (**http://www.asterisk.org/**). Asterisk is an open source framework for building communications applications.

For the BGAN terminal product, VAMs may also want to consider using Asterisk, or alternatively write their own code. Asterisk is Open Source and distributed under the **GNU General Public License** version 2 (**https://wiki.asterisk.org/wiki/display/AST/License+Information**)

G.729 Codec is a royalty-free download from SiproLab (**http://www.sipro.com/G-729.html**).

The following documents are also useful to reference:

> **RFC 3261 - SIP: Session Initiation Protocol**

This document describes Session Initiation Protocol (SIP), an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences.

> **RFC 4733 - RTP Payload for DTMF Digits, Telephony Tones and Telephone Signals**

This memo describes how to carry dual-tone multifrequency (DTMF) signalling, other tone signals, and telephony events in RTP packets.

6: Terminal Requirements

As Inmarsat is hosting the BGAN IP Voice service, VAMs only need to provide simple support on any BGAN terminal they are developing using the BRM. The terminal needs to be able to support streaming IP service.

The VAM shall implement a SIP Proxy providing the following functions for end user devices:

> **Local IP Services**
>
> The SIP proxy shall serve as a SIP Registrar for local SIP devices, e.g., VoIP application on Smartphone; wired SIP phone; etc.

> **Authentication**
>
> Local SIP devices shall be authenticated, in particular if using WLAN connections (i.e., to prevent unauthorised use).

> **2/4 Wired handsets** (optional)
>
> VAMs may want to implement one or more FXS (Foreign exchange Subscriber) interfaces for POTS handsets.

> **Transcoding**
>
> The SIP proxy should also handle transcoding to/from G.729 in order to support local SIP devices, which do not natively offer G.729.

Towards the network, the SIP Proxy needs to support:

> **PDP Context Activation**
>
> On start-up the SIP proxy shall initiate a Background PDP context for SIP signalling (dedicated APN).

> **SIP Registration**
>
> The SIP proxy then registers with the SIP Server (using IMSI as part of the SIP User Name).

> **SIP Signalling**
>
> All SIP Signalling for Incoming and Outgoing Calls.

7: SIP Client Functional Requirements

This section specifies the mandatory requirements for BRM-based equipment implementing a User Agent Client or SIP Proxy. In general, the requirements specified in this section apply to both implementations (generically referred to as SIP Client) unless indicated otherwise.

7.1: General Requirements

7.1.1: Support for Short Code Dialling

The SIP Client shall support Short Code Dialling.

7.1.2: Configuration Requirements

The SIP Client shall provide a suitable, user-friendly interface for its configuration. As a minimum, the interface shall allow the user to change the following parameters through a suitable user interface:

> Access Point Name (APN) string [*IPVOICE.BGAN.INMARSAT.COM*]

> SIP Domain Name string [*sip.bganip.inmarsat.com*]

It is recommended that these parameters are pre-configured by the manufacturer with default strings as advised by Inmarsat prior to shipping the equipment.

7.2: Functional Requirements

7.2.1: Supplementary Services Settings

The SIP Client shall allow the user to apply changes to supplementary services settings. Supplementary Service Codes applicable in the VoIP domain are specified in **VoIP Domain Call Service Codes**.

7.2.2: Protocol Requirements

Several protocol parameters are referenced in the following sections. These are shown in italics and surrounded by square brackets, e.g., [*sip_domain_name*]. Some parameters are explicitly configurable through a user interface while other are held inside the Terminal and need to be retrieved by the SIP proxy as required.

7.2.3: Session Management Requirements

In order to allow incoming and outgoing calls via the VoIP Service Domain, the SIP Client requires IP connectivity to the SIP Server and Media Gateway in the network.

7.2.3.1: Primary PDP Context Establishment

On start-up the SIP Client shall verify whether the Terminal is attached to the PS Domain and if that is the case, initiate the activation of a primary Background PDP Context towards an APN dedicated to the provision of IP transport to the VoIP Service Domain.

On successful activation of the PDP Context, the SIP Client shall retain the IP address assigned to this PDP context [*contact_ip*] for use in SIP signalling.

If the primary PDP context is deactivated for any reason, the SIP Client shall attempt to reactivate the dedicated PDP context.

7.2.3.1.1: PDP Context Management - Low Power Mode

In low power mode the BRM would deactivate the context and detach from the network. There is no way to bring the BRM back from low power mode remotely and hence the exit from low power mode shall be initiated by the host processor. When the user exits low power mode, the VoIP context would need to be re-activated by the host processor (subject to the presence of a handset or another VoIP client) as this is a PPPoE connection and is always initiated outside the BRM, by the host processor in this case.

7.2.3.1.2: PDP Context Management - PDP Context Disconnected by the Network

If the primary VoIP context is disconnected by the network, the BRM will notify the user on the Websocket and connection status API, and the client/host processor should reactivate the PPPoE connection for the VoIP. Most of the PPPoE clients offer timers and retry mechanisms when a connection is dropped and this facility could be used to retry.

Inmarsat advises that the cause of the disconnection is considered when retrying as a terminal that is mis-configured for VoIP services, e.g., incorrect credentials or APN, will get a data connection reject from the network and this could lead to the terminal repeatedly trying to activate the connection and thereby using network resources unnecessarily.

7.2.3.1.3: PDP Context Management - BRM Disconnects PDP Context Due to a High Streaming Rate

If the BRM disconnects the VoIP background connection upon a high rate streaming connection set-up, the PPPoE client on the host processor would receive a PPPoE Active Discovery Terminate (PADT) indicating that the VoIP connection is lost. The Host should poll the BRM regularly to check for the status of the high data rate connection and when the connection goes down should re-activate the PPPoE connection for the VoIP service.

7.2.3.1.4: PDP Context Management - Phone Detection

Inmarsat requires the VAM implement a checking mechanism that checks if a handset or client is connected and only then establish a PDP context. Inmarsat does not recommend creating and managing a PDP context with no client connected.

7.2.3.2: Secondary PDP Context Establishment

Secondary PDP Context Activation is requested by the network when it detects a SIP call set up.

The BRM then automatically activates the secondary PDP context and implements the correct traffic flow template to carry the RTP packets over the secondary streaming context.

7.2.4: SIP Protocol Requirements

The SIP Client shall implement a SIP User Agent as specified in **Version 2.0** of **RFC 3261 - SIP: Session Initiation Protocol**. Only UDP transport shall be used for all SIP signalling. The

PBX shall send all SIP signalling to the SIP server using UDP destination port 5060 but may select a different UDP port to receive SIP messages. If using a port number other than 5060, the PBX shall indicate its SIP port in the parameter [*port_no*] as specified below, otherwise the use of this parameter is optional.

7.2.4.1: SIP Registration

The International Mobile Subscriber Identity (IMSI) stored on the USIM within the Terminal shall be used as the SIP User Name [imsi].

After successful activation or reactivation of the primary PDP Context, the SIP Client shall send a SIP:REGISTER message to the SIP Server at the IP address resolved from the SIP Domain Name string[1].

The SIP REGISTER message shall comply with the requirements in **Section 10** of **Version 2.0** of **RFC 3261 - SIP: Session Initiation Protocol**, with *Table 1* specifying those fields which require specific values to be used.

Field Name	Field Value
From:	<sip:[imsi]@[sip_domain_name]>
To:	<sip:[imsi]@[sip_domain_name]>
User-Agent:	The field value shall identify the manufacturer and version of the SIP Client
Contact:	<sip:[imsi]@[contact_ip]:[port_no]>
Expires:	3600

Table 1. Field Values for SIP REGISTER Message

7.2.4.2: SIP Options

The SIP Client shall respond to SIP OPTIONS polling from the network.

7.2.4.3: SIP Call Setup (originating from SIP Client)

To originate a SIP call from the PBX, the PBX shall send a SIP INVITE message. In addition to the requirements in in **Section 10** of **Version 2.0** of **RFC 3261 - SIP: Session Initiation Protocol**., *Table 2* specifies those fields which require specific values to be used.

Field Name	Field Value
From:	<sip:[calling_number]@[sip_domain_name]> or <sip:[imsi]@[sip_domain_name]>
To:	<sip:[dialled_digits]@[sip_domain_name]>

Table 2. Field Values for SIP:INVITE Message

No Session Description Protocol (SDP) Offer is required in the SIP:INVITE for SIP Client originated calls. The SIP Client shall always adhere to the SDP Answer returned by the SIP Server, in particular

[1]If a valid IP address is specified instead of a SIP Domain Name string, then the IP address shall be used instead.

the ptime parameter (see **Codec Frame Packetisation**) shall be applied to the outbound RTP stream.

The SIP Client shall provide Call Progress Tones towards the calling handset and convert any incoming SIP error messages to appropriate call failure tones (or optionally, voice announcements).

7.2.4.4: SIP Call Setup (terminating on PBX)

Incoming SIP INVITEs will carry either the primary or an additional MS-ISDN as the called number in the SIP:To header in the format

```
<sip:[called_number]@[sip_domain_name]>
```

where the [*called_number*] is presented in International Number Format, i.e., with the leading 870 Inmarsat Country Code.

Incoming SIP:INVITEs will always carry an SDP offer, the SIP Client shall accept the first codec in the list as well as the ptime parameter specified. The SIP Client is not required to provide audible ringback tones towards the caller.

7.2.4.5: SIP Call Maintenance

The network may send reINVITE at regular intervals on every call to check that the SIP User Agent is still contactable and to check that the call is active in order to ensure accurate billing. The network fails the call if the SIP Client SIP User Agent does not reply or replies negatively. The SIP Client must respond positively to the re-INVITE if and only if there is already an active call matching that specified in the re-INVITE.

7.2.5: Media Handling Requirements

7.2.5.1: Codec

The SIP Client shall support G.729 or G.729A and may optionally support variant G.729B.

7.2.5.2: Discontinuous Transmission

Discontinuous transmissions, or DTX, is supported, which means that the sender does not transmit voice frames containing silence.

The G729B codec serves the same purpose as DTX, and is known as Voice Activity Detection (VAD)).

7.2.5.3: Codec Frame Packetisation

The SIP Client shall support sending multiple codec frames in a single RTP packet as determined from the SDP ptime attribute sent by the network to the SIP Client in the SIP:INVITE, SIP:183 Session Progress and SIP:200 OK messages. The following ptime values shall be supported: 20 (default), 40, and 80 ms.

7.2.5.4: DTMF Handling

DTMF Tones to and from the SIP Client shall be carried over RTP as telephone-events in accordance with **RFC 4733 - RTP Payload for DTMF Digits, Telephony Tones and Telephone Signals**.

7.2.6: SIP Domain Supplementary Services Requirements

7.2.6.1: Calling Number Presentation on Inbound Calls

For Inbound calls the Calling Number is carried in the SIP:From header in the format

```
<sip:[calling_number]@[sip_domain_name]>
```

The Calling number on Inbound Calls is normally presented in International Format with a leading "+" or "00". If the caller has withheld their number or if the number is not available for other reasons, then the [*calling_number*] parameter will contain the word "anonymous". The SIP Client should present the Calling Line ID to the handset to which the call is directed (if the handset supports Calling Line ID).

7.2.6.2: Calling Number Presentation on Outbound Calls

For Outbound calls the Calling Number is carried in the SIP:From header in the format

```
<sip:[calling_number]@[sip_domain_name]>
```

If Calling Number Presentation on Outbound Calls is selected by the user then the SIP Client shall set the [*calling_number*] parameter in the SIP:From Header.

If the calling number is not configured in the SIP Client, then the SIP Client shall use the following format:

```
<sip:[imsi]@[sip_domain_name]>
```

7.2.6.3: Supplementary Services Settings related to Calling Number Delivery

The SIP Client shall optionally allow the user to block or unblock Calling Number Delivery for the entire PBX. **VoIP Domain Call Service Codes**provides a list of applicable codes used for this purpose in the SIP Domain.

7.2.6.4: Call Waiting and Call Hold in the SIP Domain

Call Waiting and Call Hold is not supported by the Multi-Voice SIP server in the network. However, a SIP Client may support these features locally towards connected handsets.

Appendix A: VoIP Domain Call Service Codes

Call Service	Code	Confirmation
Busy Call Forwarding – enable	*90 <number to forward to>	Tone
Busy Call Forwarding – disable	*91	n/a
Calling Number Delivery – enable	*65	n/a
Calling Number Delivery – disable	*85	n/a
Calling Number Delivery Blocking – single call	*67 <called number>	n/a
Calling Number Delivery Blocking – override	*82 <called number>	n/a

8: Glossary

Term	Abbreviation	Definition
Access Point Name	APN	The name of a gateway between a GPRS, 3G or 4G mobile network and another computer network, frequently the public Internet. A mobile device making a data connection must be configured with an APN to present to the carrier.
Broadband Global Area Network	BGAN	Inmarsat's BGAN service provides simultaneous voice and broadband data communications globally from small and lightweight satcom terminals. The service is provided by Inmarsat's global constellation of I-4 satellites.
BGAN Radio Module	BRM	
Dual-Tone Multi-Frequency	DTMF	The generic term for Touch-Tone capability on a phone, which produces DTMF tones as you press the buttons.
Foreign Exchange Service	Fxs	The wall jack or the interface to the telephone system which FXO devices can be connected to. Using these interfaces a call can be established.
International Mobile Subscriber Identity	IMSI	An internationally standardized unique number to identify a mobile subscriber. The IMSI consists of a Mobile Country Code (MCC), a Mobile Network Code (MNC) and a Mobile Station Identification Number (MSIN).
Internet Protocol	IP	The method or protocol by which data is sent from one computer to another on the Internet.

Private Branch Exchange	PBX	A private branch exchange (telephone switching system within an enterprise) that switches calls between VoIP users on local lines while allowing all users to share a certain number of external phone lines.
PDP context		A **Packet Data Protocol** (PDP) context transfers information about your data connections between the BGAN terminal and the BGAN network. The PDP context defines connection aspects such as routing, QoS and security. The BGAN terminal opens a primary PDP context or a secondary PDP context, depending on the IP data connection type. Refer to BGAN and PDP contexts for details.
Real-Time Transport Protocol	RTP	
Subscriber Interface Module	SIM	
Session Initiation Protocol	SIP	A signalling communications protocol, widely used for controlling multimedia communication sessions such as voice and video calls over IP networks.
User Datagram Protocol	UDP	A communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that uses the Internet Protocol (IP). UDP is an OSI Layer-3 protocol.
Voice over Internet	VoIP	A methodology and group of technologies for the delivery of voice

Protocol	communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms commonly associated with VoIP are IP telephony, Internet telephony, broadband telephony, and broadband phone service.