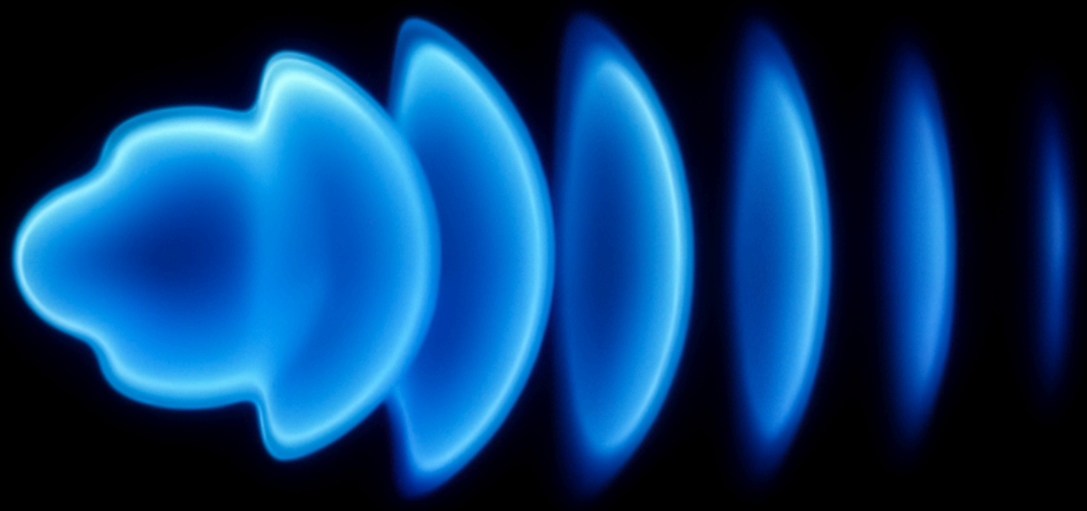
modulus

AI Voice Connectivity

Connect your AI Voice Platform
to the Public Telephone Network



Updated as of 2026

Table of Contents

Overview 02.

What We Can Provide 02.

Inbound Calls 03.

Outbound Calls 04.

Call Transfer 04.

Secure Voice Connectivity 04.

Integration Process 05.

Already Using a Voice AI Platform? 06.

Contact Us 07.

modulus provides voice services and SIP connectivity for connecting **Conversational AI, Voice AI, Contact Center, and Communication Platforms** to the public telephone network.

Through standard SIP connectivity, an external platform can use modulus services to handle **inbound and outbound telephone calls.**

What We Can Provide

Depending on the use case, modulus can provide:

- Greek telephone numbers
- Inbound calling to the AI / Communication Platform
- Outbound calling through the modulus network
- SIP Trunk connectivity
- SIP Registration
- IP-based connectivity
- SIP URI forwarding
- Secure connectivity using TLS and SRTP

Interconnection is based on standard SIP and can be used by AI Agents, Contact Centers, PBXs, and other platforms that support SIP connectivity.

Inbound Calls

Different connectivity models are supported for delivering inbound calls to the platform.

SIP Registration

The platform establishes a SIP registration with the modulus infrastructure.

Static IP Forward

Inbound calls are delivered directly to the public IP addresses specified by the platform.

SIP URI Forward

Inbound calls can be routed to a specific SIP URI.

This option provides greater flexibility for architectures that use load balancing, redundancy, or high availability.



Outbound Calls

The following authentication methods are supported for outbound calling:

- **SIP Digest Authentication,** using a username and password
- **IP Authentication,** using predefined public IP addresses

Call Transfer

Calls can be transferred from the AI platform to a human agent, allowing the agent to continue the conversation. This is typically achieved through one of the following methods:

- Conference Call
- SIP Refer

Secure Voice Connectivity

The modulus infrastructure supports:

- SIP over UDP
- SIP over TLS
- RTP
- SRTP

For secure interconnections, **TLS combined with SRTP** is recommended, where supported by the platform.

Integration Process

The initial interconnection can be completed through a straightforward process:

1. Use Case

Καταγράφουμε τις βασικές ανάγκες της πλατφόρμας και αν απαιτείται inbound, outbound ή αμφίδρομη τηλεφωνία.

2. Connectivity

The appropriate SIP connectivity method is selected.

3. Demo / Testing

A trial service is provided so that the interconnection can be implemented and compatibility between the two systems can be confirmed.

4. Technical Validation

If a specific technical blocking issue arises during testing, the technical teams can work together to resolve it.

5. Production Onboarding

Following the successful completion of testing, production onboarding can proceed.

Already using a Voice AI Platform?

If your platform supports **standard SIP connectivity,** we can assess its interconnection with modulus voice services.

For the initial assessment, we only need to know:

- the platform you are using,
- whether you require inbound, outbound, or bidirectional telephony,
- your preferred SIP connectivity method,
- and any specific technical requirements.

For detailed technical interconnection parameters, please refer to the **SIP Interconnection Technical Specifications**.



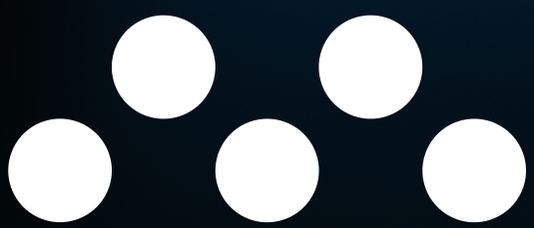
Contact Us

311, Mesogeion
Avenue, Chalandri
15231, Greece

+30 215 215 1500

telecoms@modulus.gr

For more information and comprehensive solutions,
explore our updated website: www.modulus.gr



modulus

Your Voice, Our Connection.