ABSTRACT

In the modern digital work era, the integration between legacy communication systems such as Internet Protocol Private Branch Exchange (IP PBX) and cloud-based collaboration platforms like Microsoft Teams has become a strategic necessity for many organizations. Common challenges include protocol differences, complex configurations, and the need for adequate infrastructure. This study aims to implement the integration of the Asterisk IP PBX system with Microsoft Teams using the Direct Routing method as an efficient, secure, and flexible two-way voice communication solution.

The system is developed on a virtual server hosted on Microsoft Azure, running Asterisk on Ubuntu, and utilizes the Session Initiation Protocol (SIP) and Transport Layer Security (TLS) for signaling, as well as the Real-time Transport Protocol (RTP) for media transmission. Communication between SIP clients (softphones and IP phones) and Teams users is established via a validated public Fully Qualified Domain Name (FQDN) encrypted with a TLS certificate from Let's Encrypt. Testing is conducted for two-way voice communication (SIP to Teams and vice versa).

The test results demonstrate excellent system performance, with an average delay of 19.99 ms, 0% packet loss, average jitter of 0.306 ms, and stable throughput in accordance with the TIPHON standard. The system supports up to 276 simultaneous calls before CPU utilization reaches its maximum limit of 99.4%, indicating the server's capability to handle high loads. The Asterisk configuration, acting in an SBC-like role, effectively manages signaling and media, while the Fail2Ban mechanism successfully blocks unauthorized access to SIP and SSH services. The main challenge, configuring Asterisk on Ubuntu, was resolved through independent exploration. This integration demonstrates that Asterisk can serve as a reliable open-source SBC alternative for bridging local communication systems with modern collaboration platforms such as Microsoft Teams.

Keywords: Asterisk, IP PBX, Microsoft Teams, Direct Routing, SIP, TLS, RTP, Session Border Controller, Fail2Ban, VoIP Integration