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A SIP based Soft Handoff Mechanism for Seamless Mobility in Heterogeneous Wireless Networks

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Abstract – Technical requirements for limitless network movement between different wireless systems require strong handoff protocols to accomplish uninterrupted connections. This research proposes a Session Initiation Protocol (SIP)-based soft handoff architecture as a system which delivers uninterrupted IP layer mobility management. SIP signaling in combination with media gateway implementation of packet replication along with filtering functions enables the proposed solution to achieve lower packet loss and faster handoff processes. A simulation environment designs plural wireless network technologies including GPRS, CDMA and WLAN to model the functionality of Mobile Hosts (MHs) maintained through multiple wireless interfaces during handoff activities. The system operates in real-time while updating SIP sessions and sending dynamic INVITE signals and re-INVITE messages through intelligent media gateway control points to support QoS. The performance evaluation established through key metrics including handoff latency and packet loss together with throughput and session continuity proves superior than hard handoff techniques. SIP-based soft handoff improves mobile robustness which establishes it as a suitable solution for next-generation wireless multimedia networks.

Keywords – SIP, Soft Handoff, Heterogeneous Networks, Mobility Management, Session Continuity, Packet Replication, Wireless Communication.

I. INTRODUCTION

Wireless networks are gradually progressing towards efficient and cost-effective all-IP models, interconnected with the web for worldwide coverage of services [1]. Mobile consumers anticipate access to uninterrupted real-time and non-real-time IP services at all times and locations. Robust mobility control will therefore be a crucial facilitator of this aim. Mobility management encompasses two essential functions: handoff management and location management, which correspond to the active and dormant mode of a MH (mobile host), correspondingly.

Location control may further integrate location updates at the relevant servers and the establishment of sessions for call delivery. Location updates at home mobility services is referred to as home registration. Paging of the IP may be unnecessary during the establishment of a session during which the system is monitoring the MH at subnet tier. Mobility control might be classified at intra (microdomain) and inter (macrodomain) categories. The distinction between micro mobility and macro mobility enables tailored enhancement for every specific mobility context, facilitating the delivery of the most suitable solutions. Session Initiation Protocol (SIP) and Mobile IP (MIP) have been consistent for the management of multimedia session management and IP mobility, correspondingly, but both are also possible candidates for macro-mobility control.

Literature contains illustrations of mobility control executed at the transportation layer and establishment of a distinct top mobility tier [2]. The concept of managing mobility at the application tier using SIP as a mobility control protocol seems to be the predominant focus in contemporary research. The concept of employing SIP as the application tier mobility control is prevalent throughout the research society, with references [3]. The concept of using a mix of MIP and SIP is also included in many publications [4], and [5]. To our knowledge, the incorporation of the specified policy-oriented decision model employing jitter and delay metrics from the networking layer, alongside a hybrid MIP- and SIP mobility control solution for WiMAX, WLAN, and UMTS accessibility networks, is unprecedented.

Mobile wireless technology has achieved significant appeal owing to its capacity to provide pervasive data accessibility to consumers in transit. At the moment, no one wireless networking technology can offer several mobile users simultaneously high bandwidth, low delay, and large area data services. The problem of effectively and scalablely providing network connectivity to a significant number of users is addressed by Wireless Overlay Networks [6], a hierarchical arrangement of wide-area, building-sized, and room-sized data systems. The lowest levels of an overlay network are made up of wireless cells with high bandwidth but restricted coverage. Elevated tiers in the hierarchy provide less capacity but a much broader access network. When moving between multiple network settings, a mobile device having multiple wireless network connectors can link to many networks. Many types of access methods are often included in next-generation wireless systems.

The main goal of this research involves creating and assessing a SIP-based soft handoff system to provide uninterrupted connectivity while reducing service interruptions in multitransmission systems. The research studies handoff efficiency by adopting SIP signaling with packet replication to maintain high QoS during multimedia sessions. The remaining sections of our study have been organized in the following manner: Section II describes the background of the study and reviews related works on SIP-based soft handoff system. Section III highlights the problem statement prompting the research. Data and methods have been described in Section IV to highlight the experimental setup and approach. Section V discusses the proposed architecture and emphasizes results on SIP-based soft handoff, and provide a case example on the same. Lastly, Section VI concludes the study highlighting the significance of the proposed SIP-based mobility framework in providing innovative solutions for IP layer handoff between different wireless networks.

II. BACKGROUND AND RELATED WORKS

Session Initiation Protocol (SIP) is a protocol based on messages for the management of different period. There are two fundamental SIP units: SIP servers and SIP UAs (User Agents) [7]. SIP servers may be categorized into proxy servers for registration servers and session routing for UAs registration. This study mostly examines proxy servers. In the next sections of this text, the term SIP servers will refer to proxy servers unless stated differently. One of the most prevalent session types for which SIP is employed is the phone session. This study will also examine this sort of session. In a standard SIP call period, both the callee and caller possess User Agent functions and establish the connection with the assistance of SIP servers situated along the communication channel. **Fig. 1** illustrates the SIP message sequence for starting a SIP call period.

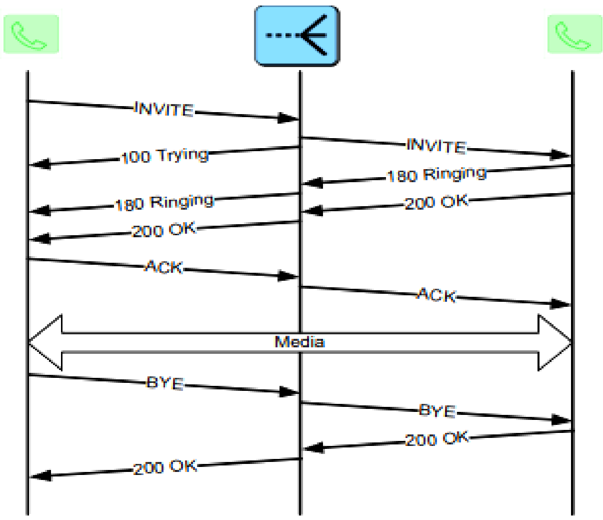

Fig 1. SIP Call Period Message Flows

The caller initiates communication by dispatching INVITE requests to SIP server that responds with a 100 Attempting response and thereafter transmits requests to the following hop identified by application-level name-based routing. In **Fig. 1**, subsequent hop for the single SIP server is callees; however, it may really be a different SIP server alongside the path. Upon receipt of the INVITE request, the callee responds with 180 Calling messages, signifying acknowledgment of callee User Agent requests, and then transmits a 200 OK response whenever callees answers the phone. The response is sent back to the callers, who will thereafter transfer ACK messages to callees to finalize the setup of the call. Subsequently, media can transmit directly between callee and the caller without the mediation of SIP servers. Whenever one side intends to terminate this call, respective UAs transmits a BYE text to the respective parties, who shall react with the OK response to acknowledge the termination of call. A standard SIP call time involves the evaluation of 5 incoming texts for call establishment and two received messages for termination of the call, resulting in all the 7 messages for the entire session.

SIP refers to the application-layer protocol that operates above the transport layer. It is capable of operating over any prevalent transport tier protocol, including TCP and UDP. A feature of SIP concerning overload issue is its timed system.

SIP has several retransmission times to address message damage, particularly when using the unstable UDP transportation [8]. We exemplify three times often associated with issues under overload conditions. Timer A initiates an INVITE resend following each expiry. Starting with the first T1 value of 500 ms, timer A escalates steadily until its cumulative timeout duration surpasses 32 s. Timer B of significance regulates the retransmission of the 200 OK responses in reaction to the INVITE requests. The countdown for 200 OK begins at T1, with its figure doubling until it attains T2 = 4 seconds. The value of the timer persists at T2 pending the cumulative timeout duration surpasses 32 seconds. Another timer of significance is Timer E, which governs the retransmission of the BYE requests. This timer adheres to a timeout pattern just like that of 200 OK timers.

The reception of matching messages generated by every original message will terminate the retransfer timer. The responses are 200 OK for BYE, ACK for 200 OK, and 100 Trying for INVITE. These responses indicates that in case an INVITE response is either discarded or remains in server line for over 500 ms without producing a 100 Trying response, the upstream SIP unit will retransfer the initial INVITE. In case the system's round-trip duration exceeds 500 ms, BYE timer, and the 200 OK timers will activate, resulting in the retransmission of these messages. In optimal network circumstances devoid of connection loss and latency, retransfers are superfluous responses that have to be circumvented.

Although the original SIP protocol did not account for end node mobility, research endeavors have been continuously undertaken to enhance mobility support in the present SIP protocol. Banerjee, Das, and Acharya [9] advocated the incorporation of mobility support into the application layer protocol SIP, where relevant, to enhance the efficiency of real-time communication. When the node relocates during the active period, it first acquires a novel address from DHCP servers (or a version thereof) and then transmits a novel session invite to the corresponding hosts see **Fig. 2**. This novel invitation specifies its novel IP address to facilitate correct packet forwarding. This invitation just serves to update the description of the current active session. Although the SIP-based technique has several benefits compared to a matching MIP-based solution, it remains afflicted by certain shortcomings. The primary disadvantage is the lack of mobility management for long-term TCP connections. Additionally, it may lead to call interruption if the novel SIP period is not fully established whereas the terminal is inside the overlapping region.

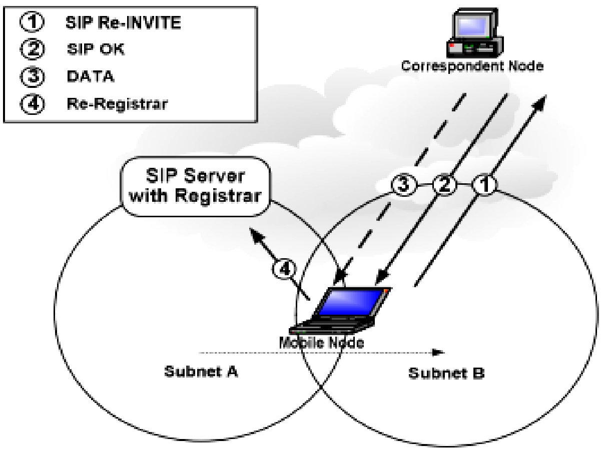

Fig 2. SIP Mobility

In contrast to a Mobile IP-based node, which may receive a Care-of Address (CoA) from a foreign agent upon detecting movement, a SIP mobility-based node must consistently obtain an IP address through DHCP. This process, contingent on application, can significantly contribute to the total handoff latency. In [10], empirical studies indicate that some standard DHCP applications produced an IP address retransmission time exceeding 2 seconds; however, the removal of DAD (Duplicate Address Detection) reduced the DHCP latency to around 0.1 seconds. Our methodology addresses the interruption time, integrating DHCP delay, associated with SIP mobility in reestablishment of calls by including mobile IP. **Fig. 3** illustrates the impact of stay time in the overlapping region and DmTOc on packet delay during handoff within SIP mobility.

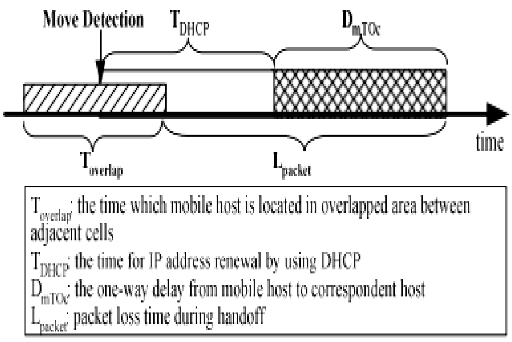

Fig 3. The Correlation Between Handoff Latency and Packet Loss

III. PROBLEM STATEMENT

Mobility is the primary characteristic of wireless systems that enables uninterrupted services in prevalent contexts. Seamless services are often accomplished via facilitating handoff [11]. Mobility management is essential for facilitating smooth traveling across diverse networks and for reducing service interruptions in real-time applications during changeover. Within a heterogeneous ecosystem, mobility-based protocols are deemed essential for facilitating global roaming across diverse access technologies. The primary methodologies for facilitating service mobility within the IP core network are MIP, which operates at the network tier, and SIP, which functions at the application layer.

Both protocols, however, exhibit distinct limitations that affect media flow during the handover process. Rapid handovers for mobility IPv6 protocol are among the suggested improvements to mobility IP within IETF mobility IP work group. Its efficiency relies on the capacity to provide two forms of handover: proactive and reactive methods. The proactive technique seeks to mitigate service deprivation that a mobile device may experience owing to a change in its point of attachment. The SIP is protocol within the application layer, which facilitates the provision of services in IP-customized systems. Consequently, it is essential to integrate fast SIP and Mobile IP to provide mobility efficiency for real-time facilities.

A mobility control protocol, functioning at the control planes and self-regulating from data planes, often facilitates handoff. As previously stated, the Session Initiation Protocol (SIP) facilitates vertical handoff management in IP-based network for media application. Whereas data plan standards provide QoS for different application, it is the role of the mobility control protocol to effectively maintain QoS through the handoff stage. Within the multimedia streaming application, the main QoS determiners are (i) delay jitter, which refers to the fluctuation in end-to-end latency among packets, (ii) end-to-end delays, and (iii) packet loss. Among these 3 characteristics, the initial two are contingent upon the network conditions along the data traffic route.

Typically, the challenges associated with these characteristics may be addressed by implementing a jitter buffer and playout. The handoff delays result in a mere glitch regarding these two determiners and has no lasting impact. Nonetheless, significant handoff delays result in substantial packet loss that severely impacts the number of media streaming applications. For example, with 16 Kbps broadcasting with 64-byte audio clusters, around 4-5 audio clusters are destroyed during a 1 second handoff delay, whilst 200,000 packets are lost or delayed for 1.50 MPEG-4 payloads with a 1050-byte packet dimension. Video quality is greatly impacted by packet loss because of error spread in MPEG-4, which particularly affects dependent framing and I-frames [12].

In voice streams, packet loss often leads to disruptive popping and clicking noises. Although SIP offers benefits in facilitating mobile support inside IP-centric heterogeneous systems, certain concerns must be addressed to provide adequate QoS assurance for media applications. SIP-based mobility handoff delay refers to the time necessary for the re-INVITE messages to traverse from the Mobile Host (MH) to the Correspondent Host (CH), albeit other processes must be executed prior to INVITE message transmission. These include: Identification of the novel networks by MH. This is contingent upon the networking initiative (e.g., intermittent beacons from accessibility points inform a mobile device of network availability in WLANs) and the OS (operating system) of the mobile host. The MH must get an IP address by a method particular to the accessibility point. This would pertain to DHCP address setting for WLAN or PDP Context Activations and Attach for GPRS networks.

A study in [13] indicates that handoff time may exceed 1 second in reduced bandwidth accessibility networks, where hard handoff, as previously discussed, significantly impacts application value. Mobility control protocol must have a way to mitigate the adverse impact of handoff delay. The soft handoff approach serves as a tool to address significant handoff delays and resultant packet loss.

IV. DATA AND METHODS

Experimental Setup

The evaluation framework included a simulation environment for assessing the SIP-based mobility framework which supports soft handoff operations at the IP layer. The infrastructure network featured a combination of wireless access systems that included Wireless Local Area Networks (WLAN) together with General Packet Radio Service (GPRS) and Code Division Multiple Access (CDMA). Various access methods supported the Mobile Host (MH) in keeping network connections active as it moved among different network zones. The Mobile Host contained multiple wireless interfaces to maintain packet transmission through various connected networks at the same time. Ensuring uninterrupted handoff and minimizing network packet loss constituted the fundamental capability of the mobile system.

Base stations served as Internet gateway endpoints when deployed inside each wireless access network for the experimental configuration [14]. The base stations operated with dual functionalities including SIP Back-to-Back User Agent (B2BUA) and SIP proxy servers to execute call control and process media packets. The B2BUA maintained session continuity through its essential control of packet filtration and duplication in SIP dialogs management. Such networks included a media gateway that served as the point of Real-time Transport Protocol (RTP) packet forwarding. MEDIA operates through two functions at once which include packet replications alongside packet filtering to avoid sending duplicate RTP packets to correct network interfaces. The simulation platform included a home registrar management service that served every MH. The managing registrar updated both their location information and network connection status for each Mobile Host. Standard SIP messages coordinated the session setup and mobility management throughout the architecture enabling MHs to update their locations automatically while crossing different network boundaries. During mobility the Correspondent Host operated as a fixed network node that both began and sustained active communications sessions with the mobile host.

Methodology
The study analyzed the efficiency level of the SIP-based soft handoff procedure through its designed methodology. The testing environment followed the path of network transition, packet replication, session re-negotiation and performance evaluation for mobile networks. A wireless network connected the MH through its active interface at the beginning of the test. The system gained a matching IP address upon crossing boundaries because it activated another interface.

During the transition the SIP User Agent (UA) inside the Mobile Host had to recognize new interface activation by initiating an INVITE message which used a JOIN header to target the previous base station. The SIP B2BUA at the base station served as the central element in directing the handoff processing. The B2BUA received the INVITE message which enabled it to identify the active session through SIP signaling dialog state parameters. The media gateway connected its packet replicator to duplicate RTP packets which directed all traffic from the old network interface to the new interface simultaneously. During this brief period the Mobile Host received duplicate data streams through both interfaces to prevent lost packets during the interface changeover. Packet filters on both the multimedia gateway and MH prevented identical RTP packets from reaching upper layers through their operation.

The MH activated re-INVITE messages toward the CH after successfully receiving packets through its new interface. Future data packets required these messages to direct them through the newly activated interface along with parameter updates. The messages which the CH received allowed it to readjust call parameters and change media packet routing paths. The new interface achieved full operational status which enabled the MH to end communication on the old interface using a BYE message. After updating its network address with the home registrar through the REGISTER message the MH could receive future session requests at the correct location.

The evaluation of a SIP-based soft handoff mechanism consisted of performance testing for handoff latency, packet loss and throughput and QoS metrics analysis [15]. Network detection time together with address configuration time and SIP signaling message transmission duration made up the handoff latency measurement. Session continuity efficiency depended heavily on the length of these delays during the assessment of the proposed solution. The examination of packet loss depended on RTP stream monitoring from time periods before the handoff event until its completion. Monitoring soft handoff operations at the IP layer showed promising results for reducing packet loss because of its superior performance compared to conventional hard handoff techniques.

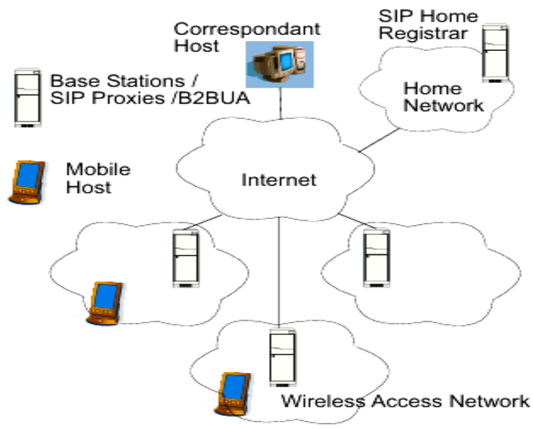

Fig 4. The Design of Next-Generation Wireless Networks

Throughput analysis was utilized during the mobility management framework assessment to measure its effects on total network procedures. The researchers measured data transfer efficiency of SIP-based soft handoff methods through packet transmission success rate evaluations. The QoS metrics allowed researchers to evaluate handoff delay effects on multimedia streaming application performance. The reliability of voice and video quality must be achieved within the proposed system strategy since these features represent direct indicators of field application success rates. Simulation tests verified the reliability of the proposed method by using different network conditions. SIP-based soft handoff functionality was examined through the research environment under varying speed scenarios and network delay along with traffic intensity levels. The simulation results established the architectural capabilities to function as future network technology for heterogeneous wireless systems.

V. PROPOSED ARCHITECTURE AND OPERATIONAL ANALYSIS

This section presents the SIP-based mobility control architecture that facilitates smooth handoff within the IP layer in diverse wireless networks. **Fig. 4** demonstrates that a mobile host (MH) may transition between diverse networks using multiple access methods, including WLAN, CDMA, and GPRS, MH is designed to interact with many access technologies and may receive and broadcast packets across several interfaces concurrently. In every wireless access network, the mobile host connects via base stations, which serve as gateways to the web. Every MH is equipped with SIP capabilities, which manage session establishment and provide smooth mobility provisioning. In accordance with the SIP design, each MH has a home system that includes an administrator service, which maintains the most current location data of MH.

The administrator function at the MH's area network is often contacted by the CH wishing to get in touch with MH in order to get the most up-to-date contact details for MH. Whenever a Mobile Host switches networks and acquires a novel IP address, as explained earlier, its SIP user begins a handoff process by transferring re-INVITE messages to the Guest Host and its local network admin service with updated SDP determinants. Handoff may also be facilitated by the BS; however, we have implemented an MH-based handoff, since it has superior awareness of the presently active network interfaces, making it the optimal choice for initiating the handoff. The handoff procedure, as described in [16], causes a significant connection disturbance, leading to packet delay. IP layer gentle handoffs have been suggested using SIP motioning to mitigate packet delay.

```
# Python-based representation of the SIP soft handoff messaging sequence

class SIPMessage:
    def __init_(self, message_type, sender, receiver, call_id, sdp_parameters=None):
        self.message_type = message_type # INVITE, JOIN, 200 OK, re-INVITE, BYE, REGISTER
        self.sender = sender # MH, CH, BS_I, BS_II
        self.receiver = receiver # MH, CH, BS_I, BS_II, Home Registrar
        self.call_id = call_id # Unique identifier for the session
        self.sdp_parameters = sdp_parameters # Session Description Protocol (SDP) data

    def __repr__(self):
        return f"{self.message_type} from {self.sender} to {self.receiver} (Call-ID: {self.call_id})"

# Example SIP message exchange during handoff

# Initial session setup between CH and MH
invite_message = SIPMessage("INVITE", "CH", "MH_Home_Registrar", "12345", "Initial SDP Parameters")
register_message = SIPMessage ("REGISTER", "MH", "MH_Home_Registrar", "12345", "MH Location
Update")

# MH detects new network and initiates soft handoff
join_message = SIPMessage("INVITE (JOIN)", "MH_UA_II", "BS_I", "12345", "Updated SDP Parameters")

# Base station assists in the transition, replicating RTP packets
ok_message = SIPMessage ("200 OK", "BS_I", "MH_UA_II", "12345")

# Re-negotiation of session parameters to update MH's new IP address
reinvite_message = SIPMessage ("re-INVITE", "MH_UA_II", "CH", "12345", "Updated Contact Info")

# Completion of handoff and session continuity assurance
bye_message = SIPMessage ("BYE", "MH_UA_I", "BS_I", "12345")

# Final registration of MH's new location
final_register_message = SIPMessage("REGISTER", "MH_UA_II", "MH_Home_Registrar", "12345",
"Updated Location")

# Printing the sequence of SIP messages
message_sequence = [
    invite_message, register message, join message, ok message, reinvite_message, bye_message,
final_register_message
]

for message in message_sequence:
```

Fig 5. Python Message Description

SIP-Centric Soft Handoff

Although the BS (base station) implements the soft handoff process, the MH initiates it [17]. Every BS has a SIP server and a SIP Simultaneously User Agent. B2BUA refers to the home entity, which accepts and executes requests as UAS (user agent servers). It preserves dialog status and engages the various requests transferred within a dialog it has initiated. All SIP communications are routed via the outbound proxies at the BS through RR record-route (RR) field within the header of the message, enabling B2BUA to effectively capture current conversation information. A multi-media entry point is linked with the B2BUA to act as a proxy and route RTP sessions. The dual functions of RTP packet separation and replication are fulfilled by multi-media entry points. MH, however, has just one filter for the packets. The replicator copies an RTP packets and transmits them to an alternate address, whereas the filter processes RTP packets obtained at the multi-media gateways and forwards a singular RTP packet copy to the end point. Media gateways and the B2BUA agent may, in theory, be manually separated from one another.

During the transition of a MH from a single network to the other, namely during handoff time, several interfaces become dynamic, enabling the MH to communicate over them. SIP imposes no restrictions on the application of the networking interface during the transmission of messages [18]. Any accessible interface may be employed by a SIP user agent to transmit feedback, a feature included in almost all SIP user applications. During the transitional phase when a novel networking interface is enabled, the SIP User Agent Client at the Mobile Host transmits INVITE feedback including the JOIN headers to Back-to-Back proxy server of the SIP user agent. For this action, the SIP user just has to be aware of the prevalent interfaces during handoff time and does not need any further assistance from networking layer.

Consequently, although soft handoffs occur at IP layers, it is wholly governed at application layers. JOIN headers typically include all pertinent information about the active call. Being an autonomous object, the B2BUA is capable of recognizing calls and then builds packets filtering mechanism and packet replicating. In order to transfer copies of every packet sent to the MH's old interface to the latest active interface, B2BUA builds the replicas at multi-media gateways. The MH uses both interfaces to send and receive packets during the temporary handoff period. The media gateway's packet filters and the MH eliminate the redundant RTP packets. Upon the arrival of packets at MH through the novel interfaces, the INVITE messages are transferred to CH, including the IP address of the novel interface and associated contact details. Resultantly, the call determiners, such as the choice of the new intermediary proxy locations and the B2BUA connected to the BS connected to the interface, are modified end-to-end.

Once the call re-negotiation is finished, as soon as a copy packet arrives at the new interface, BYE feedback is transmitted to stop the call leg over the old interface. By using REGISTER input, MH eventually updates the address data with the admin service of the local network. The example that follows demonstrates the delicate handoff method.

Example

Fig. 6-7 provide an instance of the SIP-based gentle handoff. **Fig. 5** illustrates that a period is now in continuity between the MH and the CH (correspondent host). SIP URIs CH@correspondent.com and MH@home.com indicate that MH and CH belong to distinct subnet fields. During its journey, the MH explores two distinct domains: explored_I.com and explored_II.com. BSs for these domains are BS_I and BS_II, with their Uniform Resource Identifiers being BS_I@explored_I.com and BS_II@explored_II.com, correspondingly. MH may acquire both domain-specific IP addresses using UA_I and UA_II.

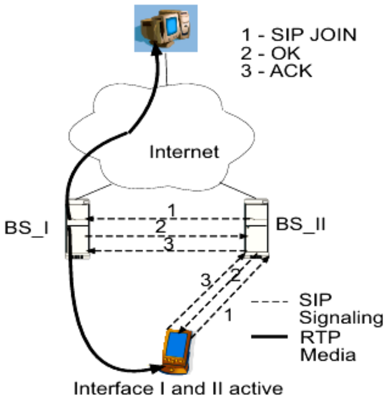

Fig 6. Dispatching of JOIN Feedback to Soft Handoff Initiation

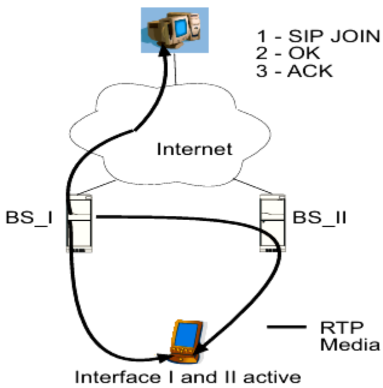

Fig 7. Division of RTP Streams-Gentle Handoff Process

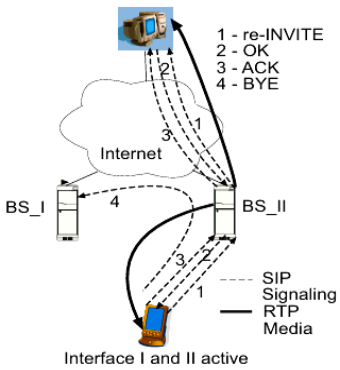

Fig 8. Notifying the Modification of the Current Session Settings Due To the Alteration of MH IP Address

Upon migrating from explored_I.com and explored_II.com, the MH activates UA II and obtains an IP address via a network-based procedure. Upon identifying the novel UA II interface, the MH SIP UA will transmit an INVITE message to BS I over this interface, including the JOIN header option. The Contact section of the INVITE feedback has the novel address for the MH: MH@explored.II.com. The SDP settings have been modified in conjunction with the new IP address. At BS I, the B2BUA utilizes from-tag, to-tag, and call-id information from the JOIN header to correlate with the current SIP dialog. BS_I constructs RTP packets repeater and ongoing dialogue filter, ensuring that UA_II gets copies of packets sent to UA_I and vice versa, together with BS. I may eliminate redundant packets sent by the MH across both interfaces. The SIP OK message is concurrently forwarded to UA II.

Consequently, for a brief duration, RTP packets arrive at both MH interfaces. While superfluous RTP packets at the multi-media ports are split and delivered to CH, those at MH are separated by encryption filters and transmitted to the upper levels see **Fig. 7**. Upon the MH's receipt of packets via UA II, it transmits re-INVITE feedback to CH to re-enter the session settings on end-to-end state, with modified endpoints. Session renegotiation clarifies the media packet route, enabling communication between the CH and MH via BS II see **Fig. 8**. Upon the arrival of a copy interface packet UA II, the link from UA I is terminated by dispatching a BYE feeback to BS I, enabling the deletion of the dialog data associated with SIP dialog via BS I. **Fig. 9** illustrates the time illustration for soft handoff, while **Fig. 5** provides a comprehensive explanation of each communication.

The handoff method comprises the following principal actions, all of which adds the delay: Address configuration and network identification operations executed by the MH [19]. The MH OS and network technologies will determine the result. sending along with the BS I JOIN header the INVITE feedback. We reconfigure the session with novel location constraints by sending the re-INVITE response. Components denoted as t_{attach}, t_{joint}, and $t_{re-invite}$ represent the corresponding delays. As mentioned before, the efficiency of service of multimedia streaming applications is adversely affected by these delays, which cause substantial packet loss. This work aims to reduce the impact of these delay elements via simple handoffs.

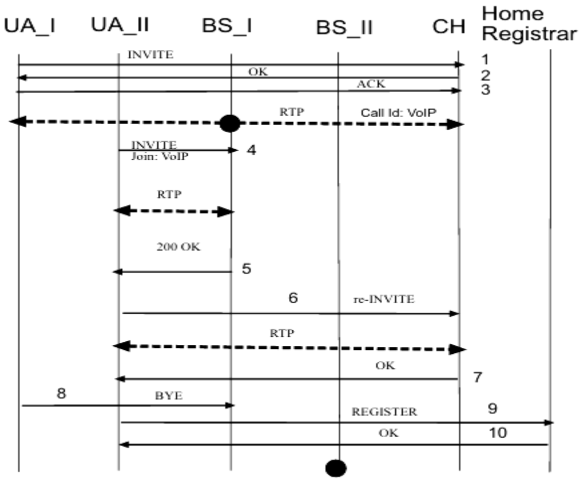

Fig 9. Message Diagram

VI. CONCLUSION

The proposed SIP-based mobility framework provides an innovative solution for IP layer handoff between varying wireless networks in next-generation networks. Using SIP signaling, the system creates continuous sessions that handle big packet loss more effectively than the usual QoS issues faced during handoff processes. The voice packet replication and filtration functionality arise from SIP B2BUA working together with media gateways during network interface transitions. The framework shows the capability to work with different access technology platforms that comprise GPRS, CDMA, and WLAN. The system framework must benefit from research-driven changes to improve multimedia streaming along with reducing handoff latency based on scientific evidence. Digital analyses through machine learning provide networks with predictive decision support for handoffs that allows them to make optimal choices within available framework boundaries. Application layer operations run independently and adapt well due to this framework since changes at lower network levels do not impact them. Users of network services are more satisfied with mobile services because SIP-based handoffs reduce interruptions when using next-generation network services at different access points.

CRediT Author Statement
The author reviewed the results and approved the final version of the manuscript.

Data Availability
The datasets generated during the current study are available from the corresponding author upon reasonable request.

Conflicts of Interests
The authors declare that they have no conflicts of interest regarding the publication of this paper.

Funding
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Competing Interests
The authors declare no competing interests.

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